2N® VoiceBlue Next

User Manual

Version 3.4
Firmware 1.16.5

www.2n.cz

The 2N TELEKOMUNIKACE joint-stock company is a Czech manufacturer and supplier of telecommunications equipment.

The product family developed by 2N TELEKOMUNIKACE a.s. includes GSM gateways, private branch exchanges (PBX), and door and lift communicators. 2N TELEKOMUNIKACE a.s. has been ranked among the Czech top companies for years and represented a symbol of stability and prosperity on the telecommunications market for almost two decades. At present, we export our products into over 120 countries worldwide and have exclusive distributors on all continents.

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2N TELEKOMUNIKACE administers the FAQ database to help you quickly find information and to answer your questions about 2N products and services. On www.faq.2n.cz you can find information regarding products adjustment and instructions for optimum use and procedures „What to do if...".

Declaration of Conformity

2N TELEKOMUNIKACE hereby declares that the 2N® VoiceBlue product complies with all basic requirements and other relevant provisions of the 1999/5/EC directive. For the full wording of the Declaration of Conformity see the CD-ROM enclosed and at www.2n.cz.

The 2N TELEKOMUNIKACE company is the holder of the ISO 9001:2009 certificate. All development, production and distribution processes of the company are managed by this standard and guarantee a high quality, technical level and professional aspect of all our

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1. Product Overview

In this section, we introduce the **2N® VoiceBlue Next** product, outline its application options and highlight the advantages following from its use. This section also includes safety instructions.

Here is what you can find in this section:

- 1.1 Product Description
- 1.2 Safety Precautions
- 1.3 Upgrade
- 1.4 Terms and Symbols Used

1.1 Product Description

2N® VoiceBlue Next is a device that helps directly interconnect a SIP–supporting VoIP network with GSM networks and can be used for direct connection with a telephone set. The voice mode, i.e. outgoing and incoming calls, is the basic function of the gateway. **2N® VoiceBlue Next** is equipped with all voice mode functions and provides the highest user comfort. In addition to voice transmission, **2N® VoiceBlue Next** enables you to send and receive text messages (SMS). No additional equipment (such as an external phone) is needed for normal operation. You can use the web interface or AT commands for all the gateway settings. The programmable parameters are set in such a way that you can make calls the moment you connect the system to the Ethernet, connect an antenna and insert the SIM card. **2N® VoiceBlue Next** can be combined with the **2N® Mobility Extension** service (remote GSM extension) for up to 8 users[1]. **2N® VoiceBlue Next** also provides connection to the **2N® SIM Star** system[2].

2N® VoiceBlue Next Basic Features

- Compact size
- Two/four GSM/UMTS channels[2]
- System rack installation option[2]
- Intelligent incoming/outgoing call routing
- SMS sending/receiving (web, AT interfaces)
- User–friendly web interface
- Power over Ethernet (PoE)
- A single antenna for 1 or 2 GSM modules
- Automatic call records and detailed statistics
- **2N® Mobility Extension** support
- **2N® SIM Star** support
- Possibility of SMS sending+receiving via web interface
- Tone detection

[1] Depending on the available licence type

[2] Depending on the part number

1.2 Safety Precautions

- It is prohibited to use any transmitters, including the UMTS/GSM devices, in areas where explosives are used, such as quarries.
- It is prohibited to use the GSM gateways at petrol stations where mobile telephones are also prohibited.
- GSM phones may affect sensitive life–saving devices in medical centres. Therefore, it is forbidden to use GSM/UMTS devices, including the GSM gateways, in such facilities.
- In general, any prohibition regarding mobile phones based on RF energy radiation applies to GSM/UMTS devices too.
- If necessary, the GSM gateways may be installed at a safe distance from the prohibited area and connected with the original place through an Ethernet cable.
- Although GSM gateways are not intended for cars or aeroplanes, all relevant prohibitions and regulations regarding mobile phones apply to them too.

1.3 Upgrade

The manufacturer reserves the right to modify the product in order to improve its qualities.

In response to the customers' requirements, the manufacturer constantly improves the software contained in the product (firmware). For the latest **2N® VoiceBlue Next** firm ware version and the User Manual refer to the 2N web sites.

For a detailed description of the **2N® VoiceBlue Next** firmware upgrade see the section devoted to the system installation.

Manual Version	Upgrade
1.1	The User Manual relates to the **2N® VoiceBlue Next** firmware version **01.00.02.**
2.0	The User Manual relates to the **2N® VoiceBlue Next** firmware version **01.00.04.**
3.0	New four–channel **2N® VoiceBlue Next** version.
3.2	The User Manual relates to the **2N® VoiceBlue Next** firmware version **1.14.0.**
3.3	The User Manual relates to the **2N® VoiceBlue Next** firmware version **1.14.0**. Text correction.
3.4	The User Manual relates to the **2N® VoiceBlue Next** firmware version **1.16.5.**

1.4 Terms and Symbols Used

Symbols in Manual

The following symbols and pictograms are used in the manual:

Safety
- **Always** abide by this information to prevent persons from injury.

Warning
- **Always** abide by this information to prevent damage to the device.

Caution
- **Important information** for system functionality.

Tip
- **Useful information** for quick and efficient functionality.

Note
- Routines or advice for efficient use of the device.

Future Functions, Innovations

The grey-marked text in this document designates the functions that are under preparation or development at present.

2. Description and Installation

This section describes the proper **2N® VoiceBlue Next** product installation and connection.

Here is what you can find in this section:

- 2.1 Before You Start
- 2.2 Factory Settings
- 2.3 Brief Installation Guide
- 2.4 IP Voice Transmission
- 2.5 VoiceBlue Next Connection to VoIP

2.1 Before You Start

Caution

- Make sure that you are equipped with all system components necessary for putting **2N® VoiceBlue Next** in operation (SIM card, VoIP phone and/or duly configured SIP line of your SIP Proxy, an available 100BaseT socket and a PC for initial settings).

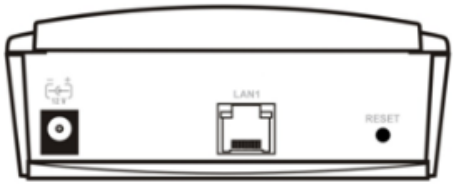
Product Completeness Check

Before installing this product, check whether the **2N® VoiceBlue Next** delivery complies with the following packing list. Be careful because there may be differences in the packing list depending on whether you have ordered a two-channel or four-channel version of the **2N® VoiceBlue Next** product (different part numbers).

Package	2ch VBN	4ch VBN
2N® VoiceBlue Next	1	1
Power supply adapter	1	1
Short antenna	1	None
Long antenna	1	1
Ethernet cable	1	1
Quick Start manual	1	1
Wall–mounting set	1	None
SIM card cover including screws	1	None
Rubber feet	None	4

Connector Lay–Out (2-Channel Version)

The following connectors are mounted on the **2N® VoiceBlue Next** bottom:
Power supply connector – DC jack 2.1mm
10/100BaseT Ethernet connector
RESET button – press for a short time to restart the GSM gateway and for a long time to restart the GSM gateway and download the factory settings.



Connector Lay–Out (4-Channel Version)

The following connectors are mounted on the **2N® VoiceBlue Next** bottom:

- 10/100BaseT Ethernet connector with PoE support
- RESET button – press for a short time to restart the GSM gateway and for a long time to restart the GSM gateway and download the factory settings.
- Antenna connector for all GSM/UMTS channels
- SIM holder

Find a power supply connector (DC Jack 2.1mm) and type plates at the rear.

Status LED Indicators

The **2N® VoiceBlue Next** states are indicated by LEDs on the front and system connector sides. For the LED types see the table below.

Power supply indicator

Blue shining	Connected, correct function
Blue flashing	Device error
No light	Disconnected, malfunction

1/2 UMTS / GSM

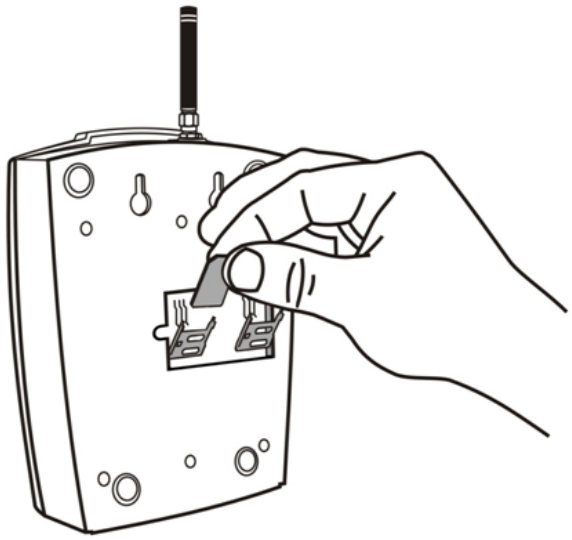
No light	Logged-in module, ready
Green shining	Active call
Green fast flashing	Module initialisation / communication
Green slow flashing	GSM/UMTS login process
Red shining	GSM module blocked /no SIM card
Red fast flashing	GSM module off
Orange shining	Initialisation upon GSM gateway power up

LAN

Green shining / Orange not shining	100BaseT Ethernet disconnected, no operation
Green shining / Orange flashing	100BaseT Ethernet connected, network operation
Green not shining/ Orange shining	10BaseT Ethernet connected, no operation
Green flashing / Orange shining	10BaseT Ethernet connected, network operation

SIM Card Placement (2-Channel Version)

Lift off the SIM card holder on the **2N® VoiceBlue Next** backside, insert the SIM card and replace the holder, securing the latch.



SIM Card Placement (4-Channel Version)

Insert a SIM card into one of the four SIM holders on the **2N® VoiceBlue Next** front making sure that the chip contacts are at the front bottom. The SIM holders are equipped with the push&pull technology for facilitation.

Caution

- Remember to **set** call forwarding, call barring, preferential network(s), SMS centre and similar provider and SIM card services in your mobile phone **before inserting the SIM card** in **2N® VoiceBlue Next**.
- If two SIM cards are used, both of them **must have an identical PIN**, or **PIN request disable** must be selected.
- **Disconnect 2N® VoiceBlue Next from the power supply** before inserting the SIM cards!

Licences

2N® VoiceBlue Next may contain time–limited software licences (SIP signalling, Mobility Extension, e.g.) that render services for limited hours only. Every gateway restart adds one hour to the internal licence counter. To check the current licence status, use the web interface of the gateway (refer to Subs. 3.4), or the AT interface of the Telnet protocol (refer to S. 4).

Caution

- A GSM gateway with an expired licence cannot process any incoming and/or outgoing calls! To avoid this, ask your dealer for licence prolongation or an unlimited licence!

GSM/UMTS Network Restriction

Some **2N® VoiceBlue Next** types may be restricted to certain GSM/UMTS networks only. If so, the red indicator at the given GSM/UMTS module is on and the 'netw-err' cause is detected. Contact your dealer please for more information.

Tip

- Contact your dealer please for more information.

Potential GSM/UMTS Troubles

All 2N GSM gateways work reliably under a long-time full load. The following problems may be caused by GSM/UMTS networks:

- GSM/UMTS module(s) cannot log in, log in slowly, or log out occasionally. This problem may be caused by any of the following situations:
  - The GSM/UMTS signal is low. The minimum signal level should be approximately **–80dBm**. If lower, change the antenna position or type!
  - The GSM/UMTS cell (BTS) to which the GSM/UMTS modules are trying to log in is overloaded. Change the antenna position or reduce the count of the logged-in GSM/UMTS modules.
- One of the GSM/UMTS modules is permanently logged-out or fails to make outgoing calls:
  - The problem indicates a GSM/UMTS network overload on the installation site. To eliminate the problem, set the **Relax delay** parameter to **2**

seconds (refer to the **GSM Basic Parameters** subsection). If the GSM module fails to log in or rejects to make outgoing GSM calls even after the gateway restart, consult your GSM provider for your SIM card/GSM module availability.

The manufacturer shall not be held liable for any SIM card or provider service unavailability in the case of a breach of the provider's SIM terms and conditions for the SIM card use.

2.2 Factory Settings

The table below includes the factory settings for the key configuration parameters of the GSM gateway:

Parameter	Value
IP address	192.168.1.2
IP mask	255.255.255.0
IP gateway	192.168.1.1
Username	Admin
Password	2n

Caution

- Remember to **change** the username and password **during** the first gateway configuration to avoid unauthorised access!

2.3 Brief Installation Guide

SIM Card Inserting/Removing

For the correct handling of SIM cards refer to the SIM Card Placement subsection.

> **Caution**
> - Make sure that the GSM gateway is off before inserting and removing SIM cards to avoid the GSM/UMTS module damage.

Installation Conditions

The following installation conditions have to be met for proper installation:

- **2N® VoiceBlue Next** is to be installed on a site with enough free space.
- **2N® VoiceBlue Next** can be mounted on a vertical surface (2-channel version), or on a horizontal surface or into a system rack (4-channel version). It is possible to operate **2N® VoiceBlue Next** in another working position too, e.g. on a desk, for a short time for servicing and testing purposes, for example.
- Any excess of the allowed working temperature may not affect the **2N® VoiceBlue Next** function immediately but may result in faster ageing and lower reliability. For the allowed working temperature and humidity ranges refer to S. 5
- **2N® VoiceBlue Next** is not designed for high–vibration environments such as means of transport, machine rooms, and similar.
- **2N® VoiceBlue Next** is not designed for dusty environments or places exposed to high humidity and temperature changes.
- **2N® VoiceBlue Next** may not be exposed to aggressive gases, acid and solvent vapours, and so on.
- **2N® VoiceBlue Next** is intended for indoor use. It may not be exposed to rain, flowing water, condensing moisture, fog, and so on.
- **2N® VoiceBlue Next** may never be exposed to direct sunshine or placed close to heat sources (radiators).
- A sufficient clearance must be kept over and under **2N® VoiceBlue Next** for cabling and air flow to carry off the heat.
- A sufficient GSM/UMTS signal intensity has to be provided for **2N® VoiceBlue Next**.
- An adequate capacity of the GSM/UMTS network has to be ensured (no BTS overload). Remember that multiple GSM gateways used in one location may overload the base transceiver station (BTS) you are currently logged in to. This may lead to a permanent or occasional rejection of GSM/UMTS calls!
- No strong electromagnetic radiation is allowed on the **2N® VoiceBlue Next** installation site.
- No strong electromagnetic reflections are allowed on the **2N® VoiceBlue Next** antenna installation site.
- An inappropriate location of **2N® VoiceBlue Next** or its antenna close to television, broadcasting and/or other rf-sensitive sets may impair the function of these sets.
- Being a source of radio frequency emissions, the **2N® VoiceBlue Next** antenna

should not occur in the close vicinity of the human body. The health hazard is higher than with mobile phones as, generally, gateways shared by multiple users show a very high traffic.

- Make sure that the VoIP connection has been configured properly according to the SIP and other VoIP recommendations.
- It is recommended that the power supply adapter should be connected to a network with a UPS back-up and due overvoltage protection.

Wall Mounting (2-Channel Version)

Follow the instructions below to wall mount **2N® VoiceBlue Next**:

1. Drill two screws into the wall in a relative distance matching the wall mounting holes in the **2N® VoiceBlue Next** rear. **The maximum screw head – wall distance is 5mm.**
2. Hang the **2N® VoiceBlue Next** unit onto the pre-drilled screws.

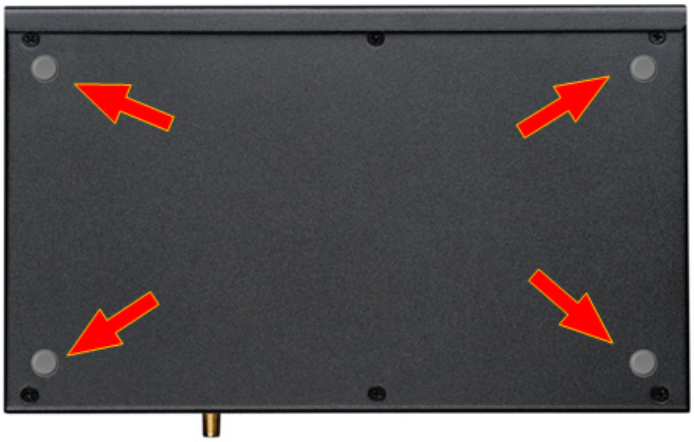
Caution

- Do not open the GSM gateway during wall mounting to avoid product damage and subsequent warranty invalidation!
- The screw head – wall distance must allow for simple handling such as removing from the wall.

Horizontal Mounting (4-Chanel Version)

2N® VoiceBlue Next is ready for mounting on a horizontal support. You can stick rubber feet (included in the delivery) on the device if necessary. Follow the instructions below:

1. Put the GSM gateway carefully on a stable horizontal support with its bottom side up.
2. Stick the rubber feet into the corners as shown below.
3. Place the device on a stable horizontal support.



caution

- Refer to the Proper Location subsection for correct placing!

System Rack Mounting (4-Channel Version)

2N® VoiceBlue Next can be mounted into a system rack. Purchase the rack accessories separately under Part No. **5051099E**. The accessory pack includes the following components:

Components	Pieces
Short wing	2
Long wing	1
Rear connecting plate	1
Upper connecting plate	1
Mounting screws	12
Rack screws	4

Single Mounting

Follow the instructions below to mount a single unit into a system rack:

1. Fit the rack wings to the right and left sides using four screws for each wing (included in the package). The positions of the long and short wings are optional

and depend on the installation site conditions.

2. Having screwed the wings, place the GSM gateway into a free 1U place of the system rack fitting it with four rack screws (included in the package).

Pair Mounting

Follow the instructions below to rack mount two GSM gateways into one 1U place:

1. Put the GSM gateways next to each other making their sides touch each other. Fit the short wings (included in the package) onto the free gateway ends with the screws enclosed.

2. Insert the connecting plates (BE CAREFUL! The rear and upper connecting plates have different drilled holes – the upper plate holes are larger and without threading) in the upper and rear holes of one GSM gateway first. Then connect the other unit.

3. Place the GSM gateway pair into a free 1U place in the system rack and fit it with four rack screws (included in the package).

Caution

- The rear and upper connecting plates have different holes – the upper plate holes are larger and without threading.
- Make sure that the plate does not get into the device to avoid electric short-circuit inside the GSM gateway.
- Leave 2 cm free space at least over and under the GSM gateway for better ventilation (airflow)!

Power Supply Connection

Use only the power supply adapter included, or, with the power over Ethernet, a certified PoE adapter to feed the gateway. Make sure that the electric distribution network voltage is in compliance with the data on the supply adapter plate before plugging the adapter. First plug the supply adapter into the mains socket and only then connect the adapter connector to the gateway. Refer to the status indicators.

Warning

- Connecting a defective or inappropriate power supply adapter may lead to a temporary or permanent **2N® VoiceBlue Next** error!
- Never connect **2N® VoiceBlue Next** using the PoE and a local adapter at the same time to avoid permanent **2N® VoiceBlue Next** malfunction!
- Check whether the antenna is connected before plugging the adapter. **Feeding the device without antenna connection may result in the GSM module transmitter damage.**

Antenna Connection

2N® VoiceBlue Next is equipped with a SMA female antenna connector for all the GSM/UMTS modules. The external antenna should always be installed vertically on a site with a good wireless signal.

Warning

- Tighten the antenna **connector gently with your hand – never use a wrench**!
- Being a source of radio frequency emissions, the **2N® VoiceBlue Next** antenna should not occur in the close vicinity of the human body. The **health hazard** is higher than with mobile phones as, generally, gateways shared by multiple users show a very high traffic.

Note

- The antenna has a sufficient gain for a trouble-free operation under normal conditions. If the signal is poor or you want to place your antenna separately from **2N® VoiceBlue Next**, you can use an antenna with an SMA–connector terminated cable. The antenna should be mounted vertically.
- Refer to the **Technical Parameters** section for the antenna and cable parameters.

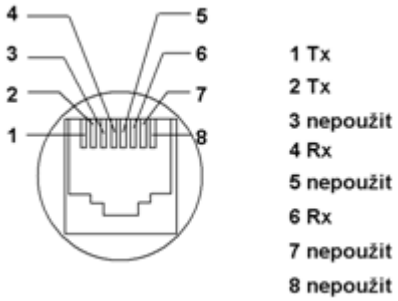
Ethernet Cable Connection

To connect **2N® VoiceBlue Next** into an Ethernet network use a standard straight cable terminated with RJ–45 connectors (included in the package). The GSM gateway supports the 10BaseT and 100BaseT standards, the Ethernet connection status is indicated by the status LED indicators located on the RJ–45 connector (refer to Subs. 2.1 for details).

For the Ethernet interface factory settings for **2N® VoiceBlue Next** refer to Subs. 2.2.

Caution

- Resetting factory values results in a change of the **2N® VoiceBlue Next** Ethernet interface configuration!
- Using a defective Ethernet cable may lead to a high packet loss rate in the Ethernet network and subsequent instability and poor quality of all GSM/UMTS calls!



RJ–45 connector for LAN connection

Antenna Splitter

The antenna splitter is a passive component that combines multiple GSM/UMTS channels into one antenna. In **2N® VoiceBlue Next**, it combines two/four antennae into one. The antenna splitter is placed in the installation box. It is a passive element – it has a characteristic signal attenuation value that the antenna connected must compensate. No antenna splitter is used for one-channel **2N® VoiceBlue Next** gatewa ys.

Licence Restrictions

2N® VoiceBlue Next may contain time limited software licences. See p. for more information.

Firmware Upgrade

Please upgrade the **2N® VoiceBlue Next** firmware before installing the system. Check the www.2N.cz web sites for the latest firmware for this gateway type.

Warning

- Use the **firmware certified** for this gateway type only! Any other firmware type may **damage 2N® VoiceBlue Next irreversibly**!

Follow the instructions below to download firmware easily using the gateway web interface:

- Connect your PC and the gateway into the Ethernet network.
- Open the web browser (MS Internet Explorer 9 and higher or Mozzila Firefox 4 and higher are recommended).
- Enter the http://IP_address to register at the web interface.
- Click on **Management–>Firmware update**, then on **Browse** and select the new firmware file.
- Click on the **Download firmware** icon in the lower part of the web page.
- **2N® VoiceBlue Next** will upgrade the firmware automatically.

2.4 IP Voice Transmission

Speech Encoding Methods

Voice transmission is strictly separated from signalling in VoIP networks. Modern VoIP networks mostly use the RTP (Realtime Transport Protocol) for voice transmission. The purpose of the RTP is only to transmit data (voice) from a source to a destination at real time. Codecs are used to save the channel data capacity. Codecs process the voice signal using variable algorithms to minimise the volume of user data. The degree of compression used by the codec affects the quality of voice transmission. Thus, the better voice transmission is required, the wider data range (the higher transmission rate) is needed. The MOS (Mean Opinion Score) scale is used for rating voice transmission quality, where 1 means the worst and 5 the best quality. For a survey of the codecs supported by **2N® VoiceBlue Next** refer to the table below.

Codecs supported

Standard	Algorithm	Transmission rate [kbps]	MOS
G.711a	PCM	64	4.1
G.711u	PCM	64	4.1
G.729*	CS–ACELP	8	3.92

[*] G.729 is an optional part of the system.

For **2N® VoiceBlue Next**, quadruple the above mentioned rates (two fully duplex calls) and add the TCP and IP header transmission rate to the result to get the resultant transmission rate.
It is important to keep both a stable appropriate transmission rate during connection and a small and identical transmission time per data packet in order to maintain a high-quality voice transmission.

- G.711 – this codec is used in digital telephone networks. The PCM (Pulse Code Modulation) is used for voice signal encoding. The sampled signal is encoded in 12 bits and then compressed using a non-linear scheme into the resultant 8 bits. Europe uses the A–law compression system while North America and Japan obey the μ-law. The resultant data flow is 64 kbps.
- G.729 – this codec uses the CS–ACELP (Conjugate-Structure Algebraic-Code-Excited Linear-Prediction) algorithm with the resultant transmission rate of 8 kbps. The speech signal is split into blocks of 10 ms each. The parameters of these blocks are then inserted in frames of the size of 10 bytes. 2–byte frames are generated for noise transmission.

During call set-up, a codec is selected automatically for voice transmission. **2N® VoiceBlue Next** supports the codecs included in the table above. The type of codec to be used depends on your VoIP network (individual devices) and your **2N® VoiceBlue Next** configuration. **2N® VoiceBlue Next** is designed primarily for VoIP corporate networks and tries to meet the opponent's codec requirements. If a codec is requested that is incompatible with **2N® VoiceBlue Next**, the call will be rejected.

The SIP and ITU–T H.323 recommended protocols are mostly used for connection establishing, maintaining and cancelling. **2N® VoiceBlue Next** uses the **SIP** (Session

Initiation Protocol) signalling.

> **Tip**
>
> - In the case of separated direct connection of your SIP Proxy and **2N® VoiceBlue Next**, use the G.711 codec to achieve a high voice quality.

SIP Components

The following components are involved in the SIP message exchange:

- UAC (User Agent Client) – the terminal device client, which initiates SIP signalling.
- UAS (User Agent Server) – the terminal device server, which responds to SIP signalling from the UAC.
- UA (User Agent) – a SIP network terminal (SIP phones, gateways to other networks, etc.), which contains the UAC and UAS.
- Proxy server – receives connection requests from the UA and transfers them to the next Proxy server if the given station is not under it administration.
- Redirect server – receives connection requests, but, instead of sending them to the called line, sends them back to the requesting device asking for where to route the request.
- Location server – receives registration requests from the UA and updates the terminal database accordingly.

All the server components (Proxy, Redirect, Location) are mostly on one physical device called Proxy server, which is responsible for keeping a client database and connection establishing, maintaining and terminating, as well as call routing.

The **2N® VoiceBlue Next** VoIP–GSM gateway acts as a UA in any case (has the same functions as a VoIP phone), i.e. receives call set-up requirements and, on the basis of its inner LCR table, routes calls to GSM networks.

None of the SIP–defined server components are integrated in the **2N® VoiceBlue Next** gateway.

SIP Signalling Messages

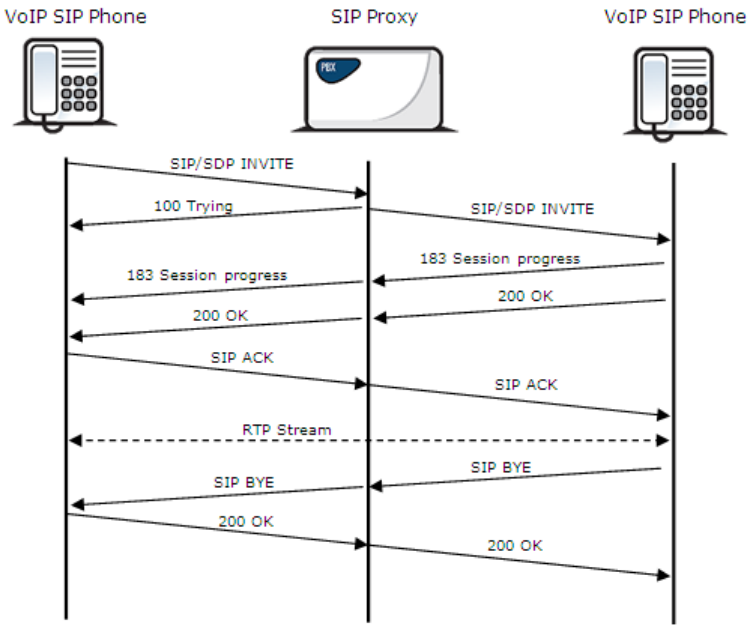
Below is a list of messages sent via the SIP:

- INVITE – connection set-up request;
- ACK – INVITE confirmation by the final message addressee;
- BYE – connection termination;
- CANCEL – failed connection cancellation;
- REGISTER – UA registration with the SIP Proxy;
- OPTIONS – server capability query.

The answers to the SIP messages are numerically coded as the case is with the http protocol. Below are the most important ones:

- 1XX – information messages (100 – trying, 180 – ringing, 183 – progress);
- 2XX – successful request completion (200 – OK);
- 3XX – request forwarding needed (302 – temporarily moved, 305 – use Proxy);
- 4XX – error (403 – forbidden, 486 – busy here);
- 5XX – server error (500 – Server Internal Error, 501 – not implemented);

- 6XX – global failure (606 – not acceptable).



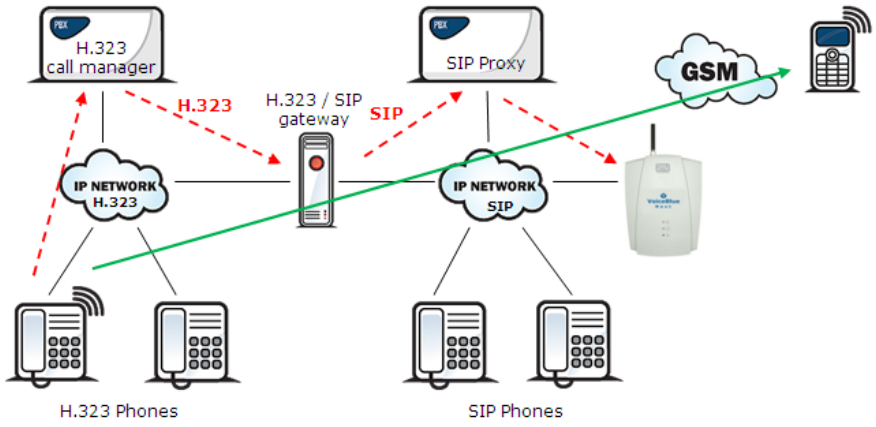
SIP Message Sending during Call Set-up and Cancellation

2.5 VoiceBlue Next Connection to VoIP

Since **2N® VoiceBlue Next** communicates using the SIP only, this subsection outlines solutions for its interconnection with networks working with the H.323 signalling protocols. **2N® VoiceBlue Next** can be used either in the Point-to-Point or Point-to-Multipoint mode with the SIP Proxy server.

SIP / H.323 Interconnection

SIP–based devices cannot communicate with H.323.supporting devices directly, but through a SIP/H.323 gateway. This gateway transfers signalling messages from/to the protocols. Using the RTP for multimedia data transmission, the SIP and H.323 protocols can go on communicating directly when the SIP/H.323 gateway connection is established. Thus, the SIP/H.323 gateway helps integrate **2N® VoiceBlue Next** into the existing H.323 environment.



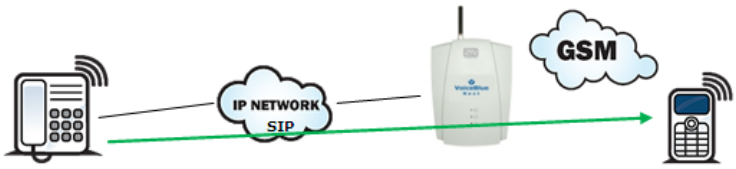
Point-to-Point Configuration

2N® VoiceBlue Next can communicate with just one SIP VoIP phone or device (a VoIP gateway, e.g.) in the PTP (Point-to-Point) mode. The IP address of the opposite party is always selected as the Proxy server IP address in **2N® VoiceBlue Next** for this mode.

The **2N® VoiceBlue Next** PTP configuration is often used for testing purposes before implementation into the VoIP network. Refer to the figure below for the PTP scheme.

Tip

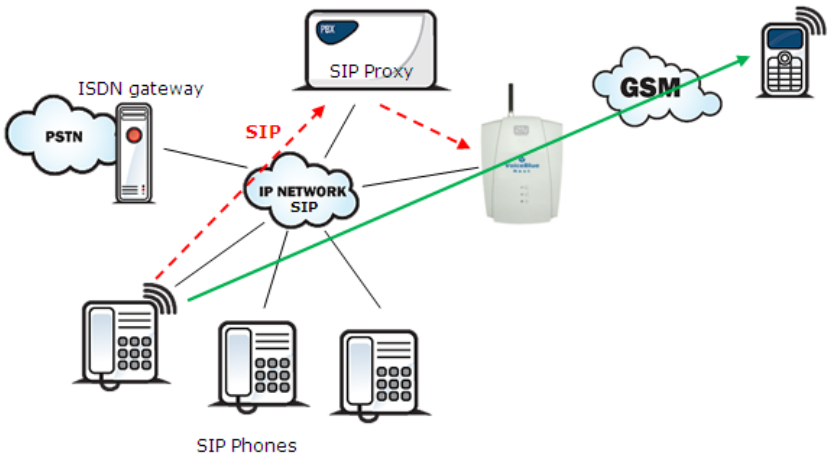
- If you set the incoming SIP Proxy IP address to '0.0.0.0' in the **2N® VoiceBlue Next** configuration, the GSM gateway will receive calls from any VoIP device.



All calls outbound to GSM are routed to the **2N® VoiceBlue Next** gateway in the Point-to-Point mode that uses **2N® VoiceBlue Next**.

Point-to-Multipoint Configuration

Point-to-Multipoint is a classical scheme of a distributed VoIP network with one or more SIP Proxy servers (VoIP gateway). The SIP Proxy server is a software PBX (or a standard PBX extended with VoIP services), which is responsible for all VoIP signalling. Multiple source devices (VoIP phones, e.g.) and multiple target devices (**2N® VoiceBlue Next**, e.g.) can be used in this mode. An internal routing algorithm (Least Cost Router, LCR) of your SIP Proxy is used for routing outgoing GSM and other calls in this mode. Calls to GSM networks can be routed via the **2N® VoiceBlue Next** gateway s. All SIP signalling is governed by the SIP Proxy server and the subsequent voice stream is based on the Point-to-Point RTP.



3. VoiceBlue Next Configuration

This section describes the **2N® VoiceBlue Next** configuration.

Here is what you can find in this section:

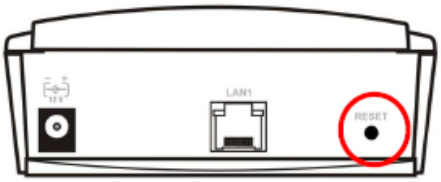
- 3.1 Factory Reset
- 3.2 Basic Configuration – Step by Step
- 3.3 Call Routing
- 3.4 Web Configuration Interface
  - Gateway control
  - Gateway configuration
  - Messaging
    - SMPP Basic Configuration – Step by Step
    - SMTP/POP3 Basic Configuration – Step by Step
  - Monitoring
    - List of SNMP traps
  - Others

3.1 Factory Reset

If you forget the password or set the IP interface incorrectly, you can press the RESET button to the right of the RJ45 Ethernet connector for a long time.
Doing this, you restore the factory default configuration values for all parameters, including the Ethernet interface parameters and access data. Refer to Subs. 2.2 for factory settings.

Caution

- Resetting the factory configuration values results in changes of the Ethernet settings and subsequent necessity to reconfigure the gateway.



Note

- Push the RESET button for a short time (0.5 s) to restart the GSM gateway.

3.2 Basic Configuration – Step by Step

This section will help put your **2N® VoiceBlue Next** gateway in operation for the first time. Refer to all S. 3 subsections for detailed settings.

- Install the GSM gateway as instructed in Subs. 2.3. Before the first start, remove the SIM cards, or insert the SIM cards with the PIN request disable.
- Connect the GSM gateway to the Ethernet network to enable connection to the address mentioned in Subs. 2.2 from the configuration terminal. If the gateway default IP address is not suitable for your Ethernet installation, reset the IP address as follows:
  - Disconnect the configuration terminal from the Ethernet network.
  - Disconnect the GSM gateway from the Ethernet network.
  - Prepare the Ethernet switches, or an Ethernet crossed cable.
  - With the Ethernet crossed cable, interconnect the configuration terminal with the GSM gateway directly.
  - With the Ethernet switch, connect the configuration terminal and GSM gateway to the pre–prepared Ethernet switch. We do not recommend connecting any other equipment.
  - Change the Ethernet IP setting in the configuration terminal, e.g.: IP=192.168.1.200, Net Mask: 255.255.255.0.
  - Open your web browser and enter the GSM gateway IP address.
  - Enter the factory login data.
  - Change the required settings in the **Gateway configuration, Ethernet configuration** menu and store the data into the GSM gateway.
  - Connect the GSM gateway to a standard Ethernet network.
  - Restore the configuration terminal IP setting and connect the configuration terminal to a standard Ethernet network.
- Enter the new GSM gateway IP address to get connected to the web interface.
- Enter the current time and date in the **Gateway control, Date/Time** menu for the GSM gateway.
- Make sure that the **Licence status** item is **unlocked** in the **Gateway control, Firmware/Licence** menu. If not, your GSM gateway does not contain the licence. Contact your dealer for the licence key.
- Set the correct PIN value in the **Gateway configuration, System parameters** menu. The value must comply with the SIM card PIN value.
- Set new login data in the **Gateway configuration, Login configuration** menu.
- Switch the GSM gateway off and insert the SIM cards. Connect an antenna to the GSM gateway and switch it on.
- The GSM gateway contains the factory configuration settings that enable outgoing calls without additional settings. Now enter the IP address equal to the GSM gateway IP address on your SIP Proxy or IP terminal.

From now on, **2N® VoiceBlue Next** will be ready to receive VoIP–SIP calls and route them to GSM/UMTS networks. If all the GSM modules are occupied, or logged out, the GSM gateway will reject all VoIP–IP and GSM/UMTS calls.
Should you get in troubles, follow the steps below please:

- Read the User Manual carefully and check all parameters.
- Find answers to the frequently asked questions at http://faq.2n.cz.
- Consult your servicing partner.

You are recommended to attend a **2N®** certified training to improve your installation chances.

3.3 Call Routing

Calls from a VoIP port to a GSM/UMTS network are routed to any GSM/UMTS port according to the LCR (Least Cost Routing) table. If an incoming call is routed via a busy port, other ports are checked automatically for availability (depending on the configuration) and in case no allowed outgoing port is available, the outgoing call is rejected.

The LCR algorithm identifies the outgoing call type, current time tariff rate, day in a week, and/or free minutes of GSM providers and routes outgoing calls accordingly.

Incoming calls from GSM networks are routed directly to the defined SIP address, or the DISA function is activated. Furthermore, calls can be routed according to the CLIP (caller's telephone number). And the CallBack service is also available.

Mobility Extension

Mobility Extension (ME) is a function that turns your mobile phone into a fixed office line and thus helps you take advantage of all PBX functions.
Advantages:

- You never miss any important call as you are available at all times.
- You can get information SMS messages on missed calls.
- You can control your company PBX call forwarding services from your mobile phone.
- You can make use of a comfortable DTMF code control.
- You need not make complicated forwarding actions as the function is fully automatic.
- ME can work with any SIP Proxy PBX.
- ME can replace any standard VoIP phone.
- Calls to your mobile phone are free of charge or at a moderate cost[4].
- You need not integrate a costly DECT system any longer.

Model Situation:

"Follow me" function

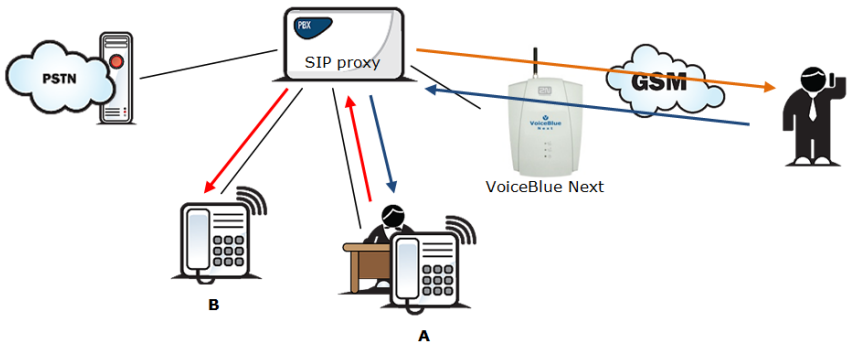


Figure 1: Follow Me Function

Fig. 1 shows routing of calls in the case of absence of a subscriber in the VoIP network. Subscriber A calls subscriber B, for whom the Mobility Extension function has been

permitted with the active "Follow me" function. Subscriber B does not answer the call in the VoIP network and so the call is rerouted to its mobile telephone.

"SMS at no answer" function

In the case of a missed call in the VoIP network, the Mobility Extension provides sending of an information text announcement. This function is called "SMS at no answer". Like with call forwarding, it is possible to use the DTMF option to activate and deactivate the service of sending text announcements for missed calls.

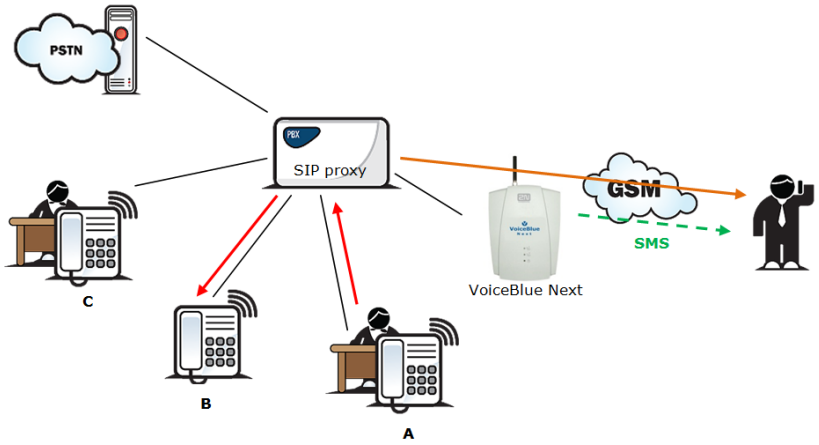


Figure 2: SMS at No Answer Function

Fig. 2 shows sending of information texts if an incoming call is not answered. Subscriber A calls subscriber B, for whom the Mobility Extension has been permitted with activated "Follow me" and "SMS at no answer" services. Subscriber B does not answer the call in the VoIP network and so the call is rerouted to its mobile telephone. Subscriber B does not answer the mobile phone call and is sent a text announcing that the call from subscriber A has been missed.

Call forwarding function

In addition to rerouting calls in the event of absence, the Mobility Extension allows calls to be forwarded within a VoIP network, which brings the services of the SIP Proxy to a mobile telephone. The description of this function is shown in Fig. 3.

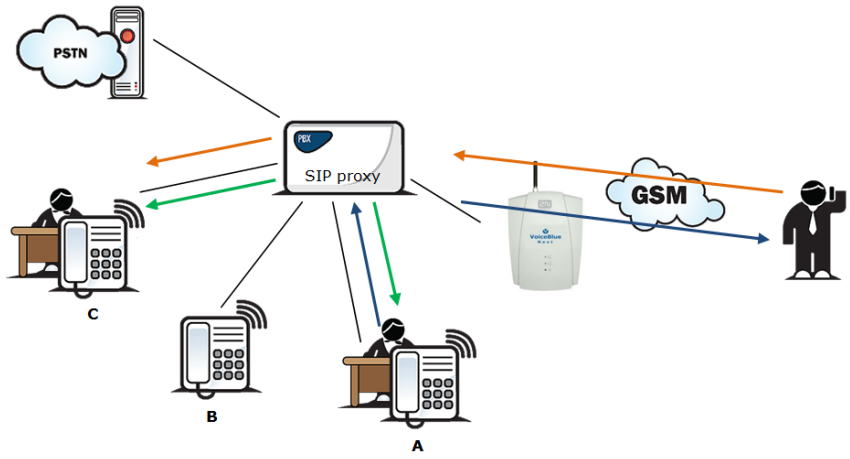


Figure 3: Call Forwarding Function

In Fig. 3 subscriber A is talking to subscriber B, for whom the Mobility Extension has been permitted. Subscriber A would like to be forwarded to subscriber C. For this reason, subscriber B holds the call with A (7* in the default setting), dials the number of subscriber C, terminates dialling with the dialling end character (# in the default setting), notifies subscriber C of the call to be forwarded and hangs up to forward the call. If subscriber C does not want to talk to subscriber A, subscriber B terminates the call with C (9# in the default setting) and returns to the call with A.

Quick call forwarding function

In Fig. 4 subscriber A is talking to subscriber B, for whom the Mobility Extension has been permitted. Subscriber A would like to be forwarded to subscriber C. Subscriber B wants to forward the call without having to talk to subscriber C. So subscriber B puts the call with A on hold (7* in the default setting), dials the quick forwarding character (default value #) and dials the telephone number of subscriber C, terminating the dialling with the dialling end character (default value #). Having received the dialling end character, **2N® VoiceBlue Next** terminates the call between A and B and attempts to make a call between subscribers A and C. Subscriber A then receives the ring tone.

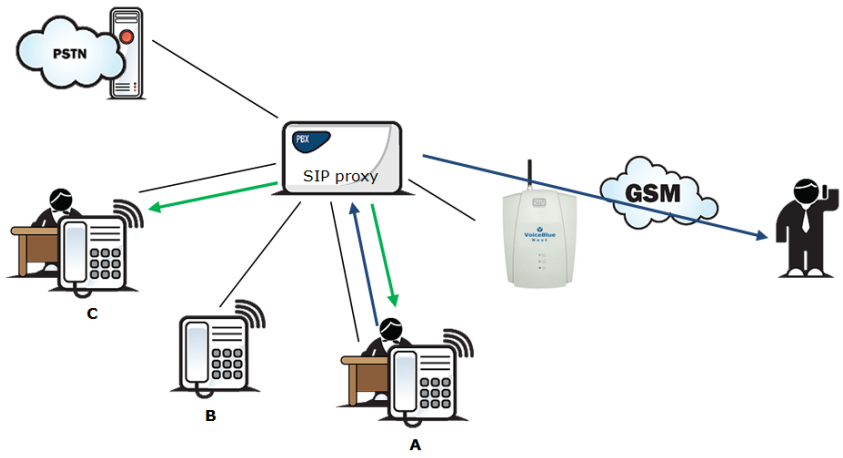


Figure 4: Quick Call Forwarding Function

Correct ME configuration:

- Connect the GSM gateway to your PBX/SIP Proxy.
- Check whether the Mobility Extension licence key has been entered correctly.
- Enter all ME users in the Gateway configuration - Mobility Extension menu.
- Enter the prefix matching the ME user mobile number in the Gateway configuration - prefixes menu.

LCR Table

The LCR (Least Cost Routing) table is the key telephone cost cutting factor. It helps you select call routes according to the called subscriber's number and the day time and weekdays. By adding bank holidays to the LCR table you achieve even higher call cost savings.

To make your prefix and LCR routing work properly, enter the prefix and the total count of digits for the number to be dialled to a GSM/UMTS network in the **List of networks**. Furthermore, make sure that the SIM card of the provider consistent with the defined group has been inserted in the GSM gateway. Use the **Assignment to GSM groups** table to assign outgoing and incoming calls to groups.

Whenever a call is to be set up, the LCR table is searched sequentially from top to bottom. If the called destination prefix matches the network prefix included in the **List of networks** table (and designated as **Network number** in the LCR table), the call is routed according to the routing group parameters (**Groups** in the LCR table) as set in the **GSM outgoing groups** table. The call will be connected via the GSM module as assigned to the **GSM outgoing group** in the **Assignment to GSM groups** table.

If the selected GSM/UMTS module is busy, the call is routed according to the next routing rule in the **Groups** as defined by the **GSM outgoing groups** table. Again, the GSM outgoing group is assigned to a GSM module in the **Assignment to GSM groups** table.

The LCR row is checked in this way until a free GSM/UMTS module is found. If no available GSM/UMTS module (**GSM outgoing groups**) is found, the call is rejected.

Routing Algorithm for Outgoing Calls

The GSM/UMTS outgoing call routing algorithm is launched whenever the SIP Proxy routes an outgoing call to **2N® VoiceBlue Next**. Outgoing calls are routed via **2N® VoiceBlue Next** as follows:

- The calling subscriber dials the subscriber number and the SIP Proxy routes it to **2N® VoiceBlue Next** depending on the SIP Proxy settings.
- The number prefix is checked against the prefixes in the **List of networks** tables in the ascending sequence, i.e. table 1 is searched first, followed by table 2 and so on.
- When a matching prefix is found in a **List of networks**, the LCR is checked for a valid row. Again, the LCR is searched upwardly, starting from the first row.
- If a **List of networks** number matches a valid number on the respective row, the call time is checked against the routing rule. If no match is found, the next LCR row is checked.
- If the **List of networks** number and call time meet the routing rules, the call is routed to the module selected in the **Assignment to GSM groups** table according to the first routing rule included in the **Groups** section and given in the **GSM outgoing groups** table.
- Moreover, call duration limiting timer can be selected for the outgoing call.
- If the selected GSM/UMTS module is occupied or has a low credit, the algorithm returns to the preceding step but considers the next **Groups** row instead of the first one. If no record is found in this section, the next LCR row is used.
- If the selected GSM/UMTS module is available and has a sufficient credit, the GSM gateway will start dialling the GSM number.
- If the calling subscriber number has an unknown prefix or all routes are

occupied, **2N®VoiceBlue Next** rejects the connection request.
- The outgoing call is not billed until the called party answers the call.
- The GSM network signals the call answering moment and the GSM gateway transfers this information to the SIP Proxy.
- It can be set for GSM outgoing calls that the calling subscriber should be sent the connection tone* instead of a silence between the request sending to GSM and the ringing tone.

* This option can be activated for TC35i modules.

Routing Algorithm for Incoming Calls

Incoming calls are processed according to the **Mode** parameter setting in the **GSM incoming groups** table. The following options are available:

- **Reject/ Ignore incoming calls** – incoming calls are not routed to the VoIP network. On the GSM side, the connection request is either rejected or ignored (the caller hears the check ringing tone).
- If the above mentioned option is not selected, the CLIP routing table is checked. If the calling number is found, it is checked for CallBack first. If the CallBack function is enabled for this number, **2N® VoiceBlue Next** will ignore the incoming call and set up a CallBack to GSM after the caller hangs up. If the AutoDial function is enabled for the calling number, the caller will be routed directly to the extension number entered in the AutoDial item. If both the CallBack and AutoDial functions are activated, **2N® VoiceBlue Next** will ignore the incoming call and set up a call to GSM after the caller hangs up. Simultaneously, a call to the VoIP extension will be set up and then the calls will be connected. If the CallBack function is enabled and the caller fails to hang up within 10 s, **2N® VoiceBlue Next** will try to set up a call according to the AutoDial settings.
- In case the CLIP routing function is disabled or the calling number is not included in the CLIP routing table, the Dynamic CLIP routing table is checked. If the calling number is found, the incoming call is routed directly to the corresponding extension. To set the Dynamic CLIP routing function use the **GSM incoming groups** menu.
- If the incoming call is still not processed, the gateway will receive the call and send either a voice message or the dialtone to the caller. After that, **2N® VoiceBlue Next** awaits the required count of digits necessary for connection set-up. Set the minimum and maximum counts of DTMF digits in the **GSM incoming groups** menu.
- If **2N® VoiceBlue Next** does not receive the minimum count of digits and no other digit comes within the timeout set in the **DTMF dialling timeout** parameter, the call is forwarded to the operator as if the called extension number were unknown.
- If call forwarding to the operator is inactive, the incoming call will be rejected.

DISA Welcome Note

If the DISA service is active and a welcome note has been recorded, the welcome note is played to every incoming call whose number is not included in the CLIP table or forwarded according to the Dynamic CLIP routing table. When the welcome note has been played, the gateway waits for the first DTMF digit for the period set in the **GSM incoming groups** – **DTMF dialling timeout** table. Having received the count of digits included in the **GSM incoming groups Minimum count of DTMF digits** table, the gateway will set up connection to the SIP Proxy. Use the GSM gateway web interface to upload the DISA welcome note.

3.4 Web Configuration Interface

Essential Data

The **2N® VoiceBlue Next** web interface supports the following web browsers:

- **MS Internet Explorer v9**
- **Mozilla Firefox v4 and higher**

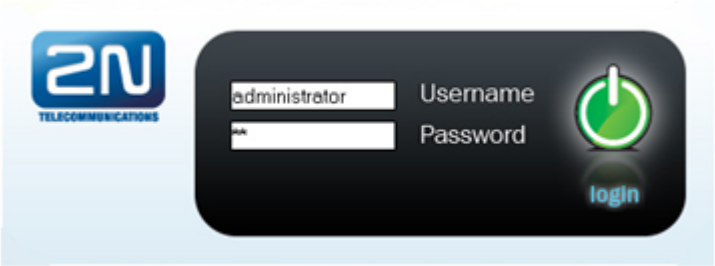
Any other web browsers may cause troubles. The recommended screen resolution is 1280x1024 and colour quality 32bit or higher. The configuration interface is available in the English language version only at present.

Tip

- Use the F11 key to activate the full–screen mode for better resolution.

Login

For login to the **2N® VoiceBlue Next** web configuration interface, enter the server IP address into your web browser. The following login dialogue will get displayed.



Just one user may be logged in at a time. Refer to Subs. 2.2 for details on the login factory settings.

Tip

- The user limitation applies to the web interface access only. The access to the Telnet interface is limited to ten simultaneous users.

A three-minute login timeout is set automatically and recovered automatically upon every user activity on the web interface. After this timeout, the current user is logged out automatically. Click on the **Refresh** button to reset the maximum timeout value. Set the login timeout value in the **Gateway–Web configuration–Auto logout** sectio n.

Caution

- You are recommended to change the initial login data upon your first login to considerably increase your system security.

Web Icons

Icon	Description
	Reset factory values
	Store current configuration into GSM gateway
	Restore previous GSM gateway configuration values
	Save onto local disk of connected terminal
	Refresh display data
	Confirmation (e.g. of configuration file storing to GSM gateway)

Caution

- Push the **Save settings** button to save the changes. Otherwise you will **lose** the configuration changes after quitting the current configuration window!

Home Page

Having logged in, you get to the **Gateway** home page as shown in the figure below. There is a menu on the left, which includes the Gateway control and Gateway configuration items. The right–hand upper corner displays the current state of the login timer and the **Refresh** button for automatic time limit refreshing.
The home page also includes the **Logout** button. You wil be notified of successful logout after every logout action to avoid re–use of your login data.
There are four more sections in the right–hand upper menu:

- **SIM client –** for connection to the **2N® SIM Star** system.
- **SMS –** for receiving/sending SMS via the web interface.

- **Messaging** – for receiving/sending SMS via SMPP or SMTP/POP3
- **Monitoring** – for gateway monitoring via SNMP
- **Utils –** including extending system tools (Network capture, Report capture).
- **Management –** for firmware update, license upload and configuration upload/download.

The main window displays information on the gateway licence status, firmware and bootware versions and the **2N® VoiceBlue Next** Ethernet interface MAC address. In addition, you can download a new licence here.

Gateway control

This group helps:

- Monitor the current statuses of the GSM gateway components;
- Check and set the GSM gateway licence;
- View and save LOG files and CDR.

Firmware / Licence

This window provides information on the gateway licensing, firmware and bootware versions and Ethernet interface MAC address. A new licence code can be inserted here.

- Firmware version – current firmware version of the gateway connected.
- Bootware version – current bootware version of the gateway connected.
- MAC address – Ethernet interface MAC address of the gateway connected.
- CPU serial number – GSM gateway serial number in the format M202–xxxxxxxxxx
- Active: licensed protocols:
  - SIP – SIP support;
  - MEx – Mobility Extension support, 'x' gives the maximum count of users;
  - G729 – G.729ab voice codec support;
  - TUN – GSM–CSD remote supervision support.
  - SMSU – count of SMS users.
  - SMSS – SMPP support.
  - SMSE – SMS@email support.
  - SMSW – SMS via web support.
  - SNMP – SNMP monitoring support.
- Gateway limitation – gateway operation time (licence limitation if any).
- Licence status – current licence status (unblocked/blocked).

Caution

- The licensed protocols will be blocked when the licence code has expired!

- Networks – list of allowed/barred GSM/UMTS networks.

Tip

- Upon the dealer's request, the gateway may contain blocking of certain GSM/UMTS networks. This state is indicated by a red shining Ch 1 / Ch 2 LED. The GSM module diagnostic window displays the 'netw–err' status.
- Contact your dealer for more information please.

- Licence key for gateway – item for entering a new gateway connecting licence.

Caution

- By entering a new licence code you restart the GSM gateway and discontinue all current calls!

Date / Time

The Date / Time window enables you to set the current date and time for the gateway. Select the **Synchronise with local PC** item and the **Time** and **Date** items will be set automatically according to your PC data.

> **Caution**
> - The internal back–up source is able to back up the internal clock source for a few hours only! Make sure that the gateway date and time values are correct after a long disconnection from the power supply!

Voice Messages

This window is used for recording, checking and downloading voice messages. Supported format is PCM–Alaw, Mono, 8000 Hz, 8 bits.

Index	Type of message	Use	Max. length(s)
0	**DISA message**	Inc. calls from GSM/UMTS	64
1	**ME "Hallo"**	Mobility Extension calls	4
2	**ME"Mobility extension"**	Mobility Extension calls	4
3	**ME"Please dial number"**	Mobility Extension calls	4
4	**ME"Text message"**	Mobility Extension calls	4
5	**ME"Activated"**	Mobility Extension calls	4
6	**ME"Deactivated"**	Mobility Extension calls	4
7	**ME"beeep"**	Mobility Extension calls	4
8	**ME"be,be,be"**	Mobility Extension calls	4
21	**GSM outgoing group 1**	Calls via Out. GSM group 1	8
22	**GSM outgoing group 2**	Calls via Out. GSM group 2	8
23	**GSM outgoing group 3**	Calls via Out. GSM group 3	8
24	**GSM outgoing group 4**	Calls via Out. GSM group 4	8
25	**GSM outgoing group 5**	Calls via Out. GSM group 5	8
26	**GSM outgoing group 6**	Calls via Out. GSM group 6	8
27	**GSM outgoing group 7**	Calls via Out. GSM group 7	8
28	**GSM outgoing group 8**	Calls via Out. GSM group 8	8
30	**Message 30**	Voice message detector	8
31	**Message 31**	Voice message detector	8
32	**Message 32**	Voice message detector	8
33	**Message 33**	Voice message detector	8
34	**Message 34**	Voice message detector	8
35	**Message 35**	Voice message detector	8
36	**Message 36**	Voice message detector	8
37	**Message 37**	Voice message detector	8

You can choose which message will be uploaded or use detection by file name. Detection requires file name: "mess[index of message][optional

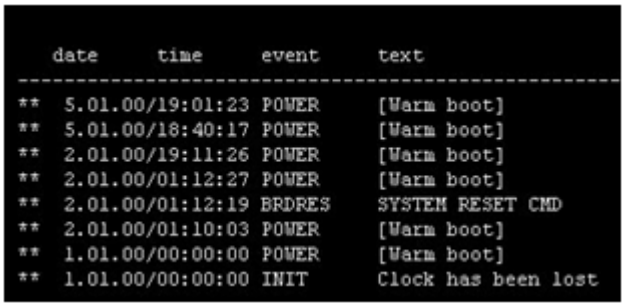
remark].wav". You can upload more than one message in .tar file.

Note

- Voice messages with indexes 30 – 37 are used for detection of the mobile provider's voice message played before call connection. If a match is found of the voice message with any of the voice messages recorded in the gateway, the call is terminated automatically or established via the last GSM outgoing group set in the LCR table (on condition that the ITD – Ignore tone detection in last group parameter is active) in the **Gateway Configuration–>LCR table** section. Refer to the **Gateway Configuration–>GSM basic parameters–>Voice message detector settings** for details.

LOG file

The LOG file window helps read out the gateway LOG file. The bottom part of the window includes icons for saving the LOG file into a file and refreshing the LOG listing in the web window.



	date	time	event	text
**	5.01.00	19:01:23	POWER	[Warm boot]
**	5.01.00	18:40:17	POWER	[Warm boot]
**	2.01.00	19:11:26	POWER	[Warm boot]
**	2.01.00	01:12:27	POWER	[Warm boot]
**	2.01.00	01:12:19	BRDRES	SYSTEM RESET CMD
**	2.01.00	01:10:03	POWER	[Warm boot]
**	1.01.00	00:00:00	POWER	[Warm boot]
**	1.01.00	00:00:00	INIT	Clock has been lost

Refer to Subs. 4.3 for more details.

CDR file

The CDR file window helps read out the Call Data Records (CDR) of the gateway. The bottom part of the window includes icons for saving the CDR into a file and refreshing the CDR listing in the web window. Refer to Subs. 4.4 for more details on the CDR format.

Caution

- The maximum capacity is 962 call records. When this limit is reached, the oldest record(s) will be deleted automatically!

Module status

This window displays the current status of each GSM/UMTS channel. Refer to Subs. 4.5 for status details.

Module control

This window helps you control the selected GSM/UMTS module manually.

Statistics

The window displays the current statistics on calls. The bottom part of the window includes icons for saving the LOG file into a file and refreshing the CDR listing in the web window. Refer to Subs. 4.6 for format information.

Current call info

The window displays the currently made calls. The bottom part of the window includes icons for saving the LOG file into a file and refreshing the listing in the web window.

Connection state

The window displays the states of all available configuration sessions. The bottom part of the window includes icons for saving the LOG file into a file and refreshing the listing in the web window.

AutoCLIP routing table

The window displays the current state of the AutoCLIP table. The bottom part of the window includes icons for saving the LOG file into a file and refreshing the listing in the web window.

Note

- The maximum AutoCLIP routing table capacity is 128 records.

SIP registration

The window displays the current SIP registration state of the gateway.

Online report

The window displays on–line gateway tracing.

Gateway configuration

System parameters

General

- Saving call data (CDR) – select the call types on which records are to be saved into the CDR file.
- Gateway ID – identifies **2N® VoiceBlue Next** numerically in the CDR in case multiple devices generate the CDR in the network.

Summer / winter time

- Automatically switch to summer/winter time – enable an automatic change of the gateway system time at the time of transition to/from the daylight saving time.
- Date of switch to winter time [dd.mm] – transition day and month.
- Date of switch to summer time [dd.mm] – transition day and month.

Mobility Extension (DTMF settings)

- Start dialling (quick call forwarding) – DTMF code for quick forwarding start.
- End dialling (quick call forwarding) – DTMF code for quick forwarding end.
- Hold call – DTMF code for active call holding.
- Hang up call – DTMF code for call termination.
- Follow me activation – activation of the **Follow me** function. The GSM gateway starts routing call to the defined GSM/UMTS subscriber number. The default value is *55.
- Follow me deactivation – deactivation of the **Follow me** function. The default value is #55.
- SMS at no answer activation – activation of the **SMS at no answer** function for a registered user. The default value is *33.
- SMS at no answer deactivation – deactivation of the **SMS at no answer** function for a registered user. The default value is #33.

Tip

- The **SMS at no answer** and **Follow me** functions can be enabled/disabled using the above mentioned DTMF codes by calling the GSM gateway from a registered mobile user phone. The values of these functions can be changed using the configuration interface too (see below).

Others

- PIN – PIN code for the SIM cards that request the PIN code.

Caution

- A SIM card with a PIN code other than that set in the GSM gateway configuration will be blocked with the 'pin-err' cause. To unblock the so-rejected SIM card, enter the correct PIN on your mobile phone!

- End of dialling (empty=off) – a selectable DTMF code for DTMF dialling end in the event of a DISA incoming call. The default value is '#'.

VoIP parameters

VoIP functions

- Day of deleting statistics on VoIP (every month) – the day of automatic deletion of call statistics on the VoIP interface. None = no deletion.

SIP protocol settings

- Use CLIP from INVITE field – the CLIP as set in the **Contact** or **From** field will be used for routing calls into GSM/UMTS networks.
- Send 180 ringing instead of 183 session progress.
- Send 200 OK instead of 180/183.
- Send 200 OK and BYE when rejected from GSM.
- Send 200 OK on REGISTER request – virtual registration of the device in **2N® VoiceBlue Next** (necessary for registration–requiring equipment).
- Replace CLIP from GSM with Caller ID.
- Deny DTMF according to RFC2833.
- Forward DTMF for ME (Mobility Extension).

SIP registration

- Registration expires [s] – the timeout after which the **2N® VoiceBlue Next** registration data expire at the SIP Proxy.
- Reattempt registration [s] – time interval for re-sending the request.
- Registration domain (realm).
- Caller ID.
- Username – registration data for the SIP Proxy.
- Password – registration data for the SIP Proxy.

Voice parameters

- First RTP port (even: 1024 – 65524) – number of the first RTP port. The number must be even as recommended.
- Last RTP port (even: first RTP+10 – 65534) – number of the last RTP port. The number must be even as recommended. The recommended minimum range of RTP ports should be 10.

Codec settings

- detailed codec settings (G.711a/u, or G.729).

Codec priority

- preferential speech codec setting:
  - Priority 1
  - Priority 2
  - Priority 3

IP addresses

- SIP Proxy (IP–>GSM) – IP address of the SIP Proxy from which **2N® VoiceBlue Next** awaits the GSM outgoing call requirements.

> **Tip**
>
> - In case you keep the default values (0.0.0.0), **2N® VoiceBlue Next** will receive requests from any IP address.

- SIP Proxy (GSM–>IP) – IP address of the SIP Proxy to which **2N® VoiceBlue Next** turns in the case of a GSM incoming call.
- SIP registrar – IP address of the SIP registration server.

> **Tip**
>
> - You can use the domain name **Registration domain (realm)** for the **SIP proxy (IP–>GSM)**, **SIP proxy (GSM–>IP)** and **SIP registrar** IP addresses on condition that you complete the domain name **Registration domain (realm)** and set the **DNS server address** properly in the **Web configuration–>Ethernet configuration** section.The **SIP proxy** and **SIP registrar** IP addresses must be set to the default value (0.0.0.0).

- NAT firewall – IP address for the NAT firewall.
- STUN server – IP address of the STUN (Simple Traversal of UDP through NATs (Network Address Translation)) server for obtaining the public IP address under which **2N® VoiceBlue Next** operates in the Internet network. You are advised to fill in this field if **2N® VoiceBlue Next** operates in a private network separated from the Internet using the NAT or firewall. The default port for sending requests to STUN is 3478.
- Next STUN request (60–6553, 0=off) [s] – used for refreshing information on the public IP address of **2N® VoiceBlue Next**. By editing this item you can configure the frequency of queries sent to the STUN server.

> **Note**
>
> - In case the GSM gateway is located behind the NAT, make sure that the NAT router routing settings for the appropriate ports (SIP, RTP, STUN) are made. The integrated firewalls may affect VoIP calls too!

> **Tip**
>
> - Should there occur call errors (one-way audibility, connection errors, e.g.), check the settings of all active elements on the VoIP call route. To detect the problems quickly, you can test the Point-to-Point connection with a software IP phone (SJ phone, e.g.) in your PC and use tracing by means of a network analyser (WireShark – www.wireshark.org e.g.) at the same time.
> - Refer to Subs. 4.2 for easy tracing through **2N® VoiceBlue Next**.

Tones generated to VoIP

- Ring tone to VoIP – generate a ringing tone of your own, or transmit a real ringing tone from the GSM/UMTS networks.

GSM basic parameters

Count of digits dialled from VoIP

- Minimum digits from VoIP – minimum count of digits to be dialled into GSM.
- Maximum digits from VoIP: maximum count of digits to be dialled into GSM.
- Wait for next digit [s] – timeout during which **2N® VoiceBlue Next** awaits further digits dialled from VoIP to GSM.

Calls

- Relax timeout [s] – time interval between the end of the last call and the beginning of the next call via one and the same GSM module (all incoming and outgoing calls are rejected during this timeout). The recommended value is 2 seconds. Do not change this setting unless absolutely necessary.
- Timeout for ringing to GSM [s] – ringing timeout for outgoing calls to GSM. If not answered or terminated within this timeout, the call will be terminated automatically by the gateway when this timeout elapses.

Bank holiday list

A list of dates to which the weekend routing mode should be applied in the LCR table.

DTMF settings

The minimum delay between two identical DTMF characters received [s/100].

Tone detector settings

The GSM gateway can automatically detect user defined tones sent by the GSM/UMTS network during call setup. In general, user defined tones are tones of the number to be transmitted. Having detected such tone, the GSM gateway terminates the call automatically and attempts to set it up through the next available outgoing group (as defined in the LCR table).

- Frequency 1;2;3;4 – defined frequency of the tone to be detected.
- Sequence list – sequence of the above defined tones for detection.

Voice message detector settings

- Minimum percent to match – set the match percentage range in which the voice message is detected as identical with one of the voice messages recorded in the **Gateway control–>Voice messages** section under index 30 - 37. The recommended value is 70-90%.

Voice parameters of GSM modules

Here you can set the voice level for the GSM modules in the GSM gateway.

Audio level DSP

Here you can set the voice level for calls in the signal processor of the GSM gateway.

- Output audio level DSP [dB] – audio volume gain/loss to VoIP
- Input audio level DSP [dB] – audio volume gain/loss to GSM

> **Caution**
> - An excessively high volume may deteriorate the voice quality (distortion, echo, etc.) and wrong DTMF detection!

Tone generated for incoming calls from GSM/UMTS

- Dialtone – the dialtone type for GSM/UMTS incoming calls.
- Ring tone – the ringing tone type for GSM/UMTS incoming calls.
- Generate busy tone to GSM/UMTS – generation of the busy tone for call end.

> **Caution**
> - If the **Generate busy tone** function is enabled, the length of the outgoing calls billed by the GSM/UMTS operator increases

Error GSM/UMTS causes

- Here you can set the ISDN release cause for the states mentioned below. The call that meets the below mentioned requirements will be rejected with a user–defined cause (the ISDN cause number will be transformed into a SIP code for VoIP according to the table shown below).
  - Lack of digits in OVERLAP mode – the call will be rejected that fails to meet the minimum digits requirement.
  - Restricted number prefix – the call will be rejected whose prefix has not been found in any of the prefix lists.
  - Selected module / GSM group is not ready – the call will be rejected in case no GSM module is available in the LCR–selected GSM outgoing group.
  - Selected module / GSM groups are not ready – the call will be rejected in case no GSM module is available in the LCR–selected GSM outgoing groups.

Cause translation

Here you can convert the release cause received from GSM/UMTS into another ISDN release cause. The resultant ISN cause number will be transformed into a VoIP SIP code as included in the table below:

Conversion table:

ISDN cause value	Description	SIP code	Description
1	Unallocated number	410	Gone
3	No route to destination	404	Not found
6	Channel unacceptable	503	Service unavailable
16	Normal call clearing	BYE	
17	User busy	486	Busy here
18	No user responding	480	Temporarily unavailable
19	No answer from user	480	Temporarily unavailable
21	Call rejected	603	Decline
22	Number changed	410	Gone
27	Destination out of order	404	Not found
28	Address incomplete	484	Address incomplete
29	Facility rejected	501	Not implemented
31	Normal, unspecified	BYE	
34	No circuit available	503	Service unavailable
38	Network out of order	503	Service unavailable
41	Temporary failure	503	Service unavailable
42	Switching equipment congestion	503	Service unavailable
44	Requested facility not subscribed	503	Service unavailable
47	Resource unavailable	503	Service unavailable
50	Requested facility not subscribed	503	Service unavailable
55	Incoming class barred within CVG	603	Decline
57	Bearer capability not authorised	501	Not implemented
58	Bearer cap, unavailable at present	501	Not implemented
63	Service or option unavailable	503	Service unavailable
65	Bearer cap, not implemented	501	Not implemented
79	Service or option not implemented	501	Not implemented
87	User not member of CVG	603	Decline
88	Incompatible destination	400	Bad request
98	Invalid message	400	Bad request
102	Recover on timer expiry	408	Request timeout
XXX	The other received CAU from netw.	500	Internal server error

Others

- Text of SMS at no answer – type the text of the SMS to be sent to the called subscriber in the event of no answer (+ the function is active). The %N string inserts the CLIP received from VoIP into the SMS.
- Text of SMS for all calls – fill in this parameter to make the GSM gateway send an SMS message to every called subscriber regardless of whether or not the call was connected. The %N string inserts the CLIP received from VoIP into the SMS.
- Save received SMS to – select the storage for SMS received.
- SIM card identification – select the SIM IMSI/SCID for CDRs.
- Disable CLIP from GSM/UMTS to VoIP – enable/disable resending the CLIP from GSM to VoIP.
- Reject call with CHLD – reject incoming GSM/UMTS calls by means of AT+CHLD (user busy) instead of standard ATH.

GSM group assignment

You can assign the GSM/UMTS modules to groups separately for incoming and outgoing calls. See the two items below for outgoing and incoming group settings.

GSM outgoing groups

2N® VoiceBlue Next allows you to work with two groups of outgoing calls for each of which you can set variable connection set–up modes and count of used minutes and sent SMS messages for a selected period.

General settings

- Delay for CONNECT [s] – define a delay before sending information on the connected call after receipt from GSM.
- Minimum ring duration to send SMS at no answer [s] – set the minimum ringing time for an outgoing call to GSM/UMTS before the SMS at no answer is sent.

> **Note**
> - The **SMS at no answer** function works properly only if the INVITE message contains the called and caller numbers.

- Delay for ALERTING [a] – define a delay before sending information on ringing start.
- Minute parameter – select whether or not the GSM gateway should record the call length or count for outgoing call restriction.
- Day of deleting statistics in group (every month) – define a day on which statistics on disconnected calls should be deleted.
- Generate virtual ring tone – enable/disable generation of the virtual ringing tone to the VoIP interface.
- Call length counting: select whether the call should be counted in seconds or minutes.
- BTS lock – identify the BTS to which the GSM modules should be logged fixedly. Restart the selected GSM modules to execute the changes.

Caution

- The BTS lock service works with specific GSM modules only (Q55)!
- An error in BTS identification results in a GSM module login failure.

- After call relax delay – interval between the current call termination and next call setup via the same GSM/UMTS module. 2 seconds is recommended for high-traffic installations.

Disconnect call

Specify the reasons for an immediate disconnection of an outgoing GSM/UMTS call.

Send CLIP from VoIP to GSM/UMTS

- Transfer CLIP to GSM/UMTS – enable/disable the function.
- Separating char – the CDN / CLIP separating character.
- Modify (" removes one digit) – you can change the CLIP. The "– character is used for deleting one char from the left.

Caution

- The **Send CLIP from VoIP to GSM** service must be supported by the GSM/UMTS provider. If not, the provider's network may reject the call!

GPRS activation

- **APN string** – define the Access Point Name (APN) for GPRS connectivity.

Basic settings

- Roaming enabled for network code – set the international network identifier for roaming purposes consisting of the following two codes:
  - MCC – Mobile Country Code – country code (Czech Republic – 230);
  - MNC – Mobile Network Code – GSM code (T-Mobile – 01, 0202, Vodafone – 03).

Hence, the international identification code for T-Mobile CZ is 23001. Leave this field empty to disable roaming.

String	Note
<empty>	Roaming is disabled
2300	Roaming is disabled (5 digits is the minimum count)
23002	Roaming is enabled for networks with MCC+MNC code 23002
230XX	Roaming is enabled for networks with MCC+MNC codes 23000 – 23099
XX001	Roaming is enabled for networks with MCC+MNC codes 00001 – 99001
XXXXX	Roaming is enabled for any network

Note
- Check the GSM/UMTS searching priorities on the SIM card using your mobile phone before enabling roaming.

Caution
- Calls via a roaming network may increase you telephone bill!

- CLIR – fill in this parameter to define whether or not the called subscriber should be displayed the calling SIM card telephone number. It is recommended that the telephone number of the SIM card inserted in the GSM module should not be displayed to the called subscriber to avoid problems with CallBack from GSM.

Caution
- The **Temporary CLIP** and **Temporary CLIR** enable services must be supported by the GSM/UMTS provider. If not, not, the provider's network may reject the call!

- Maximum count of called minutes – set the maximum count of minutes to be used via the SIM card per month.
- Count of SMS messages – set the maximum count of SMS messages to be sent via the SIM card per month.
- Day to restore call limit and delete statistics – set the day in a month on which the statistics of the **Maximum count of called minutes** a **Count of SMS messages** parameters should be deleted.
- First count – set the length of the first pulse after which the pulse counting will be changed as set in the **Next count** parameter.
- Next count – set the length of one pulse in seconds after the time as defined in the **First count** parameter elapses.

Note
- A correct setting of the **First count** and **Next count** parameters guarantees a correct counting of remaining minutes on SIM cards. These parameters help limit outgoing calls according to free minutes. The CDR includes real data.

- Day limit of called minutes – set the maximum count of minutes to be called via the SIM card per day.

Time limits

Define the SIM card use time limits for the GSM group.
Call tariffs – use this function to assign up to four independent free minute counters to each GSM module (SIM card) group:
Tariff 1 = free minutes for calls to own GSM/UMTS network;
Tariff 2 = free minutes for calls to other GSM/UMTS networks;
Tariff 3 = free minutes for calls to the PSTN;
Tariff 4 = free minutes for calls within a closed user group (VPN).
Set the LCR table (assign prefixes to tariffs) correctly to ensure full functionality of this feature. If routing to a specific tariff is not applied in the LCR table, global free minutes will be used.

- **Free minutes** Count of free minutes for a specific tariff offered by the provider.

- **Transferred minutes** Maximum count of unused free minutes to be transferred to the next period.
- **Day of restoring free minutes** Day on which the free minute counters are to be reset automatically. Choose every 24 hours, a day in a month, or a day in a week for restoring.
- **Week of restoring free minutes in month** Define the week in which the free minute counters are to be reset automatically. This setting is useful when counter reset takes place on 'every other Friday in month'. Otherwise, leave the 'every' value.

Caution

- We recommend you to set the free minute counter value to X–5, where X is the count of free minutes obtained from the GSM/UMTS provider to avoid limit excesses.
- The manufacturer is not liable for any additional call costs in case the GSM/UMTS provider's free minute/SMS limits are exceeded.

GSM incoming groups

2N® VoiceBlue Next allows you to work with two groups of incoming calls for each of which you can set variable connection set-up modes.

General settings

- Mode – set how the gateway should process incoming GSM calls:
  - Reject incoming calls – all incoming GSM calls are automatically rejected.
  - Ignore incoming calls – all incoming GSM calls are ignored. The calling subscriber gets the check ringing tone.
  - Accept incoming calls + voice message – incoming GSM calls are received and, if preset so, the DTMF dial-in with a voice welcome note is activated.
  - Accept incoming calls + dialtone – incoming GSM calls are received and, if preset so, the DTMF dial-in function with a simulated secondary dialtone is activated.
  - CallBack after ring / Reject – in case the CallBack table includes the CLIP, a CallBack is made. CLIP–less incoming calls are rejected.
  - CallBack after ring / Ignore – in case the CallBack table includes the CLIP, a CallBack is made. CLIP–less incoming calls are ignored.
  - Report to PC + voice message – the GSM gateway sends information on incoming calls to a PC equipped with the call routing software. The DTMF dial-in with a voice welcome note can be activated for incoming calls.
  - Report to PC + dialtone – the GSM gateway sends information on incoming calls to a PC equipped with the call routing software. The DTMF dial-in with a simulated secondary dialtone can be activated for incoming calls.
- Minimum digits in DTMF – set the minimum count of digits to be required by the gateway for DTMF dial-in.
- Maximum digits in DTMF – set the minimum count of digits to be accepted by the gateway for DTMF dial-in.
- DTMF dialling timeout [s] – set the time for the GSM gateway to wait for the first / next DTMF digits. If you set this parameter to 0, the incoming call is automatically connected to the numbers included in the **List of called numbers**.
- Day of deleting GSM incl. group statistics – define a day in a month on which the statistics on incoming calls should be deleted.
- Prefix before DISA dial-in – enter a numerical prefix to be automatically inserted before the DTMF dial-in.
- CLIP – use this parameter to modify the CLIP from GSM/UMTS networks. '+' is removed automatically from international identifiers. Use '–' to remove a digit. Examples (GSM CLIP +420600123456):

Parameter	CLIP to VoIP/PRI 1	Note
Null	420261301500	No CLIP change
+	+420261301500	+ inserted before CLIP received
00	00420261301500	00 inserted before CLIP received
–	0261301500	First two digits removed from CLIP received
–––99	99261301500	First three digits removed from CLIP received, prefix '99' added

- Looping of voice message – set the voice message playing time.

Send CLIP from GSM/UMTS to VoIP

- Transfer CLIP from GSM/UMTS – enable/disable the function.
- Separating char – enter the character to separate the calling SIM card CLIP and ID extension to be called.
- Modify – modify the ID extension.

> **Caution**
> - The **Send CLIP from VoIP to GSM** service must be supported by the GSM/UMTS provider. If not, not, the provider's network may reject the call!

Others

- Time to keep CLIP in table – set the time for which records are to be kept in the AutoCLIP routing table.
- Add record only for unconnected call – unconnected outdoing calls are stored in the AutoCLIP table only.
- Delete record for connected answer – if the CallBack is successful, the corresponding AutoCLIP record will be deleted.
- Skip DTMF for numbers not in CLIP Routing table – enable this function to define that only the incoming calls whose CLIPs are stored in the CLIP Routing table will be able to use DTMF dial-in.
- Skip list of called numbers after failed call to wanted number – enable this function to define that the incoming call will not be forwarded to the number(s) included in the List of called numbers in case the call is rejected after the preceding DMTF dial-in.
- Auto end to CallBack request – any incoming call whose CLIP matches the CallBack setting will be rejected and all the other incoming calls will be ignored. In either case, the CallBack function will be invoked after the incoming call ends.

List of called numbers

List of numbers to be dialled if DTMF dial–in was not made. Search the table from top to bottom. If the called user is inaccessible, use the following table record.

Prefixes

The window helps you adapt the gateway to calling to variable GSM networks. You can set the prefix-based call routing rules here and identify up to eight different groups for routing.

Prefix list 1-8

Eight prefix groups to be assigned in the LCR table.

Basic settings

- GSM network ID – name the prefix list for convenience.
- Default count of digits – the default length of the number to be dialled into GSM/UMTS when routing via the selected prefix list. Use this parameter if the **Digits count** parameter is not included in the **Accepted prefixes** table.

Table of replaced prefixes

This table helps you change the prefix of the number received (from 00 to +, e.g.). You can just add or remove a prefix using this table. This change is performed before the prefix table search is initiated. Keep the „"/ „" record at least in the table for a proper function.

Table of accepted prefixes

A list of prefixes of called destinations to which the selected prefix list applies.

Note

- The maximum count of Accepted Prefixes table records is 128.

LCR table

The LCR (Least Cost Routing) table helps route calls according to the called number and depending on the day time and weekday.

- Prefix list – select a list of prefixes to be used for the selected LCR row.
- Time limitation – set a time limitation for a particular LCR row.
- Weekend usage – enable/disable a particular PCR row during weekends.
- Maximum call length – set the maximum length (in minutes) for an active outgoing call to GSM/UMTS.
- Groups – set the GSM outgoing groups to be used for the selected LCR row.
- Tariffs – select the tariff group (free minute counter) to be used for the outgoing call. Refer to the **GSM outgoing groups** for more details.
- Ignore tone detection in last group – having detected a user defined tone (refer to **GSM basic settings**), the GSM gateway terminates the call and seeks for another call setup option by default. If this parameter is activated, the GSM gateway ignores the tone detection results and sets up the call if this is the last call setup option.

Note

- You are recommended to read the **Routing Algorithm for Outgoing Calls** section on p. before setting the LCR table and prefix lists.
- Remember to set the tariffs correctly in the **GSM outgoing groups** section to make tariff routing work properly.
- The maximum count of LCR table records is 64.

CLIP routing table + CallBack

The table permanently assigns CLIPs to the extensions to which incoming calls are to be routed automatically and set the CLIP list for the GMS network for which the CallBack function is enabled.

- GSM number (CLIP) – user identification in a GSM/UMTS network.
- AutoDial – enable/disable CLIP routing for the selected CLIP.
- Dial to VoIP – enter the VoIP destination number for CLIP routing.
- Auto CallBack – enable/disable CallBack for the selected CLIP.
- Limit – set the maximum call length (in minutes).

Note

- Remember to activate the CallBack mode in the **GSM incoming groups** window for a proper CallBack function.
- The maximum count of CLIP routing table records is 96.

Mobility Extension

The table registers the Mobility Extension users.

- Name – enter the user name for calling to VoIP.
- User – enter the registration user name for VoIP.
- Password – enter the registration user password for VoIP.
- GSM number (CLIP) – enter the user SIM card CLIP.
- Follow me function – enable/disable call forwarding to a mobile user station (based on CLIP).
- SMS at no answer function – enable/disable sending of information SMS on missed calls.

Note

- You are recommended to read the Mobility Extension section on page carefully before using the Mobility Extension function.

Ethernet configuration

This window helps you configure the Ethernet interface of your gateway.

- Use DHCP – enable/disable the DHCP client function in **2N® VoiceBlue Next**.
- IP address – the fixed IP address (v4) of the **2N® VoiceBlue Next** Ethernet interface.
- Subnet mask – the subnet mask for the **2N® VoiceBlue Next** Ethernet interface.
- Default gateway – The IP gateway IP address (v4) in the Ethernet network.
- **DNS server1** – primary DNS server.
- **DNS server2** – secondary DNS server. Used as a back-up when DNS server 1 is not functional.

Caution

- Incorrect settings, e.g. enabling the **DHCP client** service, may result in a denial of access to the **2N® VoiceBlue Next** configuration. To avoid this, restore the GSM gateway default settings. Refer to Subs. 3.1 for details.

Tip

- In case the gateway is in the DHCP client mode, current assignments obtained from the DHCP server are displayed in the **IP address**, **Subnet mask** and **Default gateway** items.

Login configuration

The window helps you set the access password and name for the **2N® VoiceBlue Next** web interface. Use the same data for Telnet connection too.

> **Caution**
> - Remember to **change** the username and password **during** the first gateway configuration to avoid unauthorised access!

Web configuration

Set additional parameters for the gateway web interface.

- Auto logout – set the count of minutes in which the current user should be logged out automatically.
- Enable web session lock – by default, one Admin user may only be connected to the GSM gateway at a moment. If another authorised user tries to log in, the previous session will be aborted automatically. By activating this function you disable such automatic logout and block any further access attempts.
- Simple login form – here you can change the login window graphic lay-out. Upon activation, the GSM gateway login page will have an anonymous appearance. This function is recommended for connecting the gateway to the Internet directly.
- Use SMS user for SMS operations on the web – enable/disable the user authorised for SMS sending/receiving only.
- SMS user name/password – the gateway enables independent connection of a user authorised for SMS receiving/sending only.

Report configuration

This window is used for setting details on automatic tracing generated by the GSM gateway.

Messaging

This group helps:

- Set the GSM gateway to send/receive SMS via SMTP/POP3 or SMPP
- Define the routing rules for incoming and outgoing SMS messages
- Check the GSM gateway SMS database filling capacity

> **Caution**
> - This service is file-licensed. Refer to **Management–>Licence key** for the current GSM gateway licence details.

Modules Grouping

Global settings

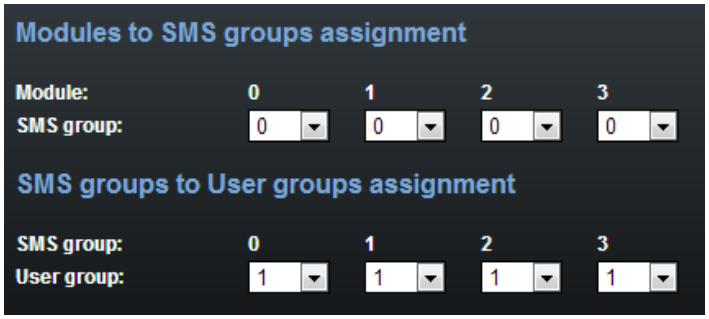
- **SMS server enabled –** enable/disable the SMS server at the GSM gateway (keep disabled if you select another SMS processing method).
- **AutoCLIP routing expires [hours] (1-10000, 0=off) –** set the maximum time for keeping of the automatic SMS routing record (response) in the AutoCLIP table.

Modules to SMS groups assignment

This section helps you assign GSM modules to SMS groups (according to the GSM provider, e.g.) to facilitate incoming/outoging SMS routing.

SMS groups to User groups assignment

This section helps you assign SMS groups to User groups for SMS sending/receving purposes. The GSM gateway can contain up to four User groups.



> **Caution**
> - The count of user accounts is file-licensed. Refer to **Management–>Licence key** for the current GSM gateway licence details.

Users

This section helps you manage the SMS users. Select a communication protocol for user adding (SMTP/POP3 or SMPP).

Add user

- **User group –** assign the user to a User group.
- **User full name –** define the system user name.
- **Replace dial for outgoing SMS –** define the number (CPN) to which the user's outgoing SMS messages shall be sent.
- **Access type –** select the communication protocol for SMS processing:
  Login name (without domain) – define the login name for the client Email.
  - **Email (SMTP/POP3)** – use SMTP for SMS sending and POP3 for SMS receiving.
  - **SMPP (SMSC)** – set the GSM gateway into the 'SMS centre' mode.
  - **SMPP (ESME)** – set the GSM gateway into the 'ESME - External Short Messaging Entity' mode.
- **System ID –** SMSC user ID.
- **Password –** SMSC/SMTP user password.
- **System ID (for authentication on remote side) –** ESME user ID.
- **Password (for authentication on remote side) –** ESME user ID.
- **TCP type –** set the TCP parameter into the client/server mode.
- **Server address –** set the server IP address and TCP server transport port.
- **Message direction of Esme –** set the ESME user details:
  - **Transceiver –** ESME works as a transmistter/receiver.
  - **Transmitter –** ESME works as a transmitter.
  - **Receiver –** ESME works as a receiver.
- **Address range –** set this parameter if required so by SMSC for SMS receiving.
- **Delivery status required** – set the SMS sending request for outgoing SMS delivery confirmation.
- **Send message as Flash SMS** – activate the Flash SMS function for outgoing SMS messages.

Note

- You can set the **Delivery status required** parameter only if the SMTP/POP3 protocol is used.
- With SMPP, the delivery information is active automatically in case the service is required by the SMPP counterparty or via an incoming SMS message from the GSM/UMTS network.

User Group

This section helps you edit the User group and set the routing rules for the group's incoming and outgoing SMS.

General settings

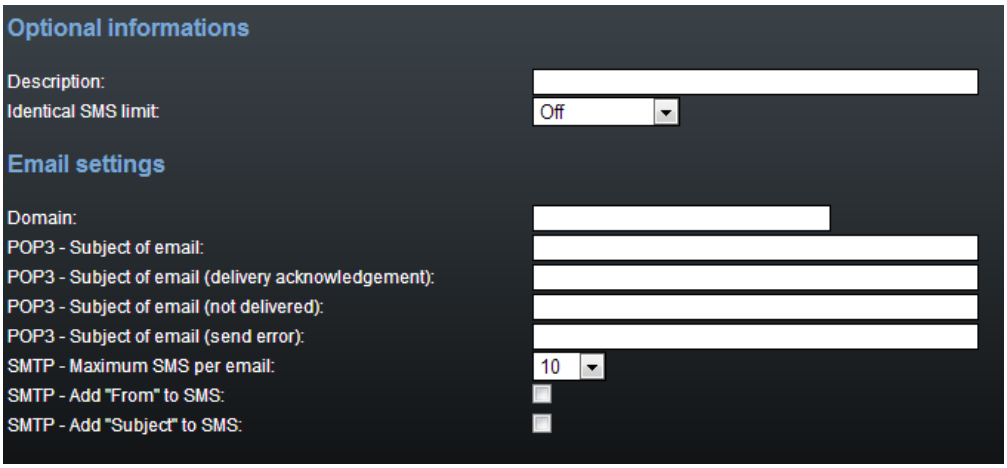
Set the basic parameters for SMS sending via SMTP/POP3 or SMPP.

Optional informations

- **Description –** set the User group details.
- **Identical SMS limit** – set the check of identical SMS messages, i.e. sequentially sent messages with identical SMS PDUs. When the set limit is exceeded, the excessive SMS messages are discarded without being sent.

Email settings

- **Domain –** define the domain name for SMS sending/receiving via SMTP/POP3.
- **POP3 - Subject of email –** set the message subject to be added automatically to the SMS loaded from the GSM gateway via POP3.
- **POP3 - Subject of email (delivery acknowledgement)** – set the SMS delivery acknowledging message subject**.**
- **POP3 - Subject of email (not delivered)** – set the SMS non-delivery acknowledging message subject.
- **POP3 - Subject of email (send error)** – set the SMS sending failure message subject.
- **SMTP - Maximum SMS per email –** set the maximum count of SMS messages to be sent within one Email message. Any characters exceeding the set length will be ignored.
- **SMTP - Add "From" to SMS –** add 'From' from the Email message to the outgoing SMS. The sender information will be displayed at the beginning of the SMS message.
- **SMTP - Add "Subject" to SMS –** add 'Subject' from the Email message to the outgoing SMS. The subject information will be displayed at the beginning of the SMS message.



SMPP address modification

- **Source address of message incoming from SMPP** – set how to modify the message sender ID received via SMPP.
    - **Never replace original address**
    - **Replace if the original address is empty**
    - **Always replace original address**
  - **address** – enter sender's new ID
  - **TON** – type of number
  - **NPI** – numbering plan indicator
- **Destination address of message incoming from SMPP** – set how to modify the message addressee telephone number received via SMPP.
    - **Never replace original address**
    - **Replace if the original address is empty**
    - **Always replace original address**
  - **address** – enter addressee's new telephone number
  - **TON** – type of number
  - **NPI** – numbering plan indicator
- **Source address of message outgoing to SMPP** – set how to modify the message sender ID to be sent via SMPP.
    - **Never replace original address**
    - **Replace if the original address is empty**
    - **Always replace original address**
  - **address** – enter sender's new ID
  - **TON** – type of number
  - **NPI** – numbering plan indicator
- **Destination address of message outgoing to SMPP** – set how to modify the message addressee telephone number to be sent via SMPP.
    - **Never replace original address**
    - **Replace if the original address is empty**
    - **Always replace original address**
  - **address** – enter addressee's new telephone number
  - **TON** – type of number
  - **NPI** – numbering plan indicator

Incoming SMS routing

This section helps you set the incoming SMS routing rules, which are included in the incoming SMS routing table.

Add record

- **Disabled –** enable/disable the selected routing rule.
- **Message from SMS group –** select the SMS group that receives the SMS.
- **Action type –** select the action for the received SMS:
  - **Autoclip routing** – the incoming SMS will be routed according to the AutoCLIP table records.
  - **Deliver to user** – the SMS will be delivered to the selected user.
  - **Drop message** – the SMS will be dropped.
- **Deliver to user –** select the SMS addressee. This option is valid only if **Action type = Deliver to user** is selected.
- **Time limitation –** set the rule validity time.
- **Days limitation –** set the days in a week on which the rule shall be valid.
- **Clip mask –** set the filter for the SMS sender telephone number (CLIP). If the selected mask matches the CLIP, the SMS will be routed according to the rules set above. If not, the SMS will be dropped. The ***** character replaces any string of characters (789789789=789*, e.g.). The **?** character replaces any character (789789789=789??????, e.g.).
- **Text begin –** set the string to be matched against the SMS text beginning. If a match is found, the SMS will be routed according to the rules set above.

Note

- The rules are applied to incoming SMS sequentially, starting from the first rule. If all the set rules are met in a row, the SMS is routed according to the row.

Outgoing SMS routing

This section helps you define the outgoing SMS routing rules, which are included in the outgoing SMS routing table.

Add record

- **Disabled –** enable/disable the selected routing rule.
- **Message to SMS group –** select the SMS group via which the SMS will be sent.
- **Action type –** select the action for the outgoing SMS:
  - **Submit from user** – enable sending SMS from the selected user.
  - **Submit from all users** – enable sending SMS from all the SMS group users.
- **Submit from user –** select the user in case **Action type = Submit from user** is selected.
- **Time limitation –** set the rule validity time.
- **Days limitation –** set the days in a week on which the rule shall be valid.
- **Dial mask –** set the filter for the SMS addressee telephone number (Dial). If the selected mask matches the Dial, the SMS will be routed accroding to the rules set above. If not, the SMS will be rejected and deleted. The * character replaces any string of characters (789789789 = 789*, e.g.). The ? character replaces any character (789789789 = 789??????, e.g.).
- **Text begin –** set the string to be matched against the SMS text beginning. If a match is found, the SMS will be routed according to the rules set above.

Note

- The rules are applied to outgoing SMS sequentially, starting from the first rule. If all the set rules are met in a row, the SMS is routed according to the row.

Control

This section helps you monitor the system. It includes storage filling information and AutoCLIP records.

SMS queues

The subsection displays the current SMS storage state.

Incoming SMS database

The subsection displays current information on the incoming SMS storage filling percentage and maximum count of incoming storage records.

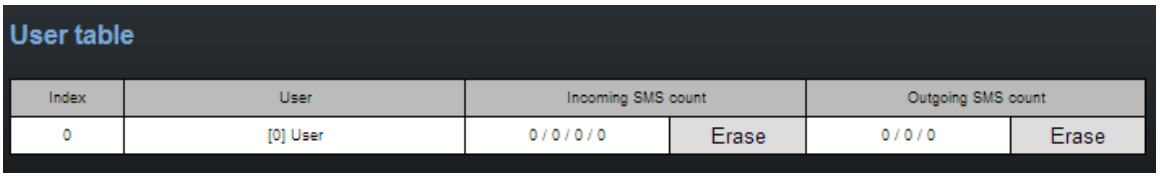
> **Caution**
>
> - The POP3 protocol enables you to set how long a copy of the loaded message shall be kept in the SMS server. The SMS keeping time depends on your e–mail client setting. Make sure that you have chosen a correct time value to avoid database overfilling. When the storage has been filled up, no new SMS messages will be received!

Outgoing SMS database

The subsection displays current information on the outgoing SMS storage filling percentage and maximum count of outgoing storage records.

User table

The table shows the count of incoming and outgoing SMS stored in the above-mentioned databases from the user's point of view and helps you delete the SMS stored.



User table items:

- **Incoming SMS count** – Received / Ack OK / Ack error / Failed
  - **Received** – count of received common SMS messages
  - **Ack OK** – count of SMS delivery acknowledging messages
  - **Ack error** – count of SMS non-delivery acknowledging messages
  - **Failed** – count of SMS sending failure messages (failed due to network error, invalid number, etc.)
- **Outgoing SMS count** – Waiting / Sending /Sent
  - **Waiting** – count of waiting SMS messages
  - **Sending** – count of currently sent SMS messages
  - **Sent** – count of sent SMS messages waiting for delivery acknowledgement

AutoCLIP routing table

The window displays the current state of the AutoCLIP table.

SMPP Basic Configuration – Step by Step

This section helps you define the basic parameters for SMS sending/receiving via SMPP in **2N® VoiceBlue NEXT**. Read the Messaging subsection carefully before setting details.

Figure 1 shows sending/receiving SMS via SMPP. Having created user accounts in **2N® VoiceBlue NEXT**, the users can send/receive SMS via SMPP. The figure shows an example with two SMSC accounts created in **2N® VoiceBlue NEXT** (to the right). Another **2N® VoiceBlue NEXT** (ESME) gateway and a PC on which the SMPP (ESME) application is running are connected to these accounts via SMPP. An SMS coming to **2N® VoiceBlue NEXT** via SMPP is routed according to the gateway outgoing routing rules and sent to GSM. An SMS coming from GSM is routed according to the gateway incoming routing rules and sent to the destination via SMPP.

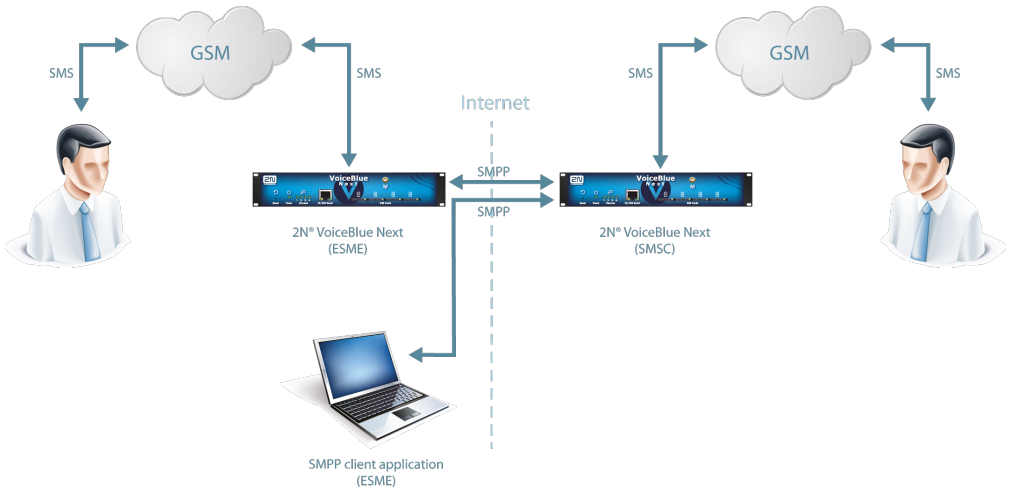


Figure 1: SMPP Communication

Take the following basic steps in **2N® VoiceBlue NEXT** to enable SMS sending/receiving:

1. In the **Monitoring–>SMS grouping** menu:
   a. Assign the selected GSM modules to the SMS groups (according to the GSM provider and SIM cards with identical tariffs, for example).
   b. Assign the SMS groups created in step 1.a above to the User groups if you use messaging with different routing rules to different company departments, for example.
2. Add users in the **Monitoring–>Users–>Add** menu:
   a. Assign the user to a **User group**.
   b. Set **Access type: SMPP (SMSC).**
   c. Set **System ID** and **Password** to access the GSM gateway SMS server via ESME.
   d. Set **TCP type: server**.
3. Create the incoming SMS routing rules in the **Messaging–>User Group–>General settings–>Incoming SMS routing–>Add** menu:
   a. Select the group to receive the SMS via the **Message from SMS group** menu.
   b. Select the **Action type** to define how to process incoming SMS. Example: Set **Action type: Deliver to user** to deliver SMS to selected users.

c. Set **Deliver to user** to to define the SMS receiving user.

4. Create the outgoing SMS routing rules in the **Messaging–>User Group–>General settings–>Outgoing SMS routing–>Add** menu:
   Enable the SMS server in the **Messaging–>Modules grouping–>SMS server enabled** menu.
   a. Set the **SMS group** via which the outgoing SMS will be routed in the **Message to SMS group** menu.
   b. Select the **Action type** to define how to process outgoing SMS. Example: Set **Action type: Submit from user** to send SMS from selected users.
   c. Set **Submit from user** to define the SMS sending user.
5. Should you get in troubles, follow the steps below please:

- Read the User Manual carefully and check all parameters.
- Find answers to the frequently asked questions at http://faq.2n.cz.
- Consult your servicing partner.

You are recommended to attend a 2N certified training to improve your installation chances.

SMTP/POP3 Basic Configuration – Step by Step

This section helps you define the **2N® VoiceBlue NEXT** basic parameters for SMS sending/receiving via SMTP/POP3. Read the Messaging subsection carefully before setting details.

Figure 1 shows sending/receving SMS via SMTP/POP3. Having created user accounts in **2N® VoiceBlue NEXT**, the users can access the accounts via the e-mail client and send SMS messages to **2N® VoiceBlue NEXT** via SMTP. The gateway receives the messages and resends them to the GSM users according to the predefined routing rules.A message coming to **2N® VoiceBlue NEXT** from GSM is routed according to the incoming SMS rules and delivered to the user via POP3.

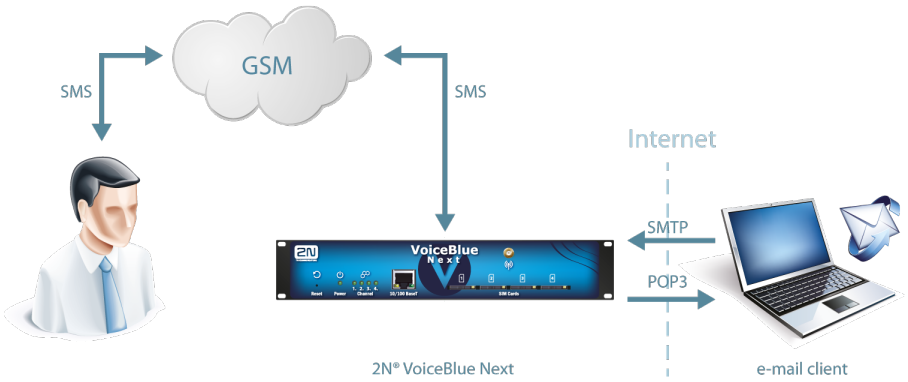


Figure 1: SMTP/POP3 Communication

> **Note**
> - You can send an SMS message of any length to up to 100 users at the same time from the e-mail client. The maximum SMS count per e-mail is only limited by the **User group–>General settings SMTP–>Maximum SMS per email** settings.

Take the following basic steps in **2N® VoiceBlue NEXT** to enable SMS sending/receiving:

1. In the **Monitoring–>SMS grouping** menu:
   a. Assign the selected GSM modules to the SMS groups (according to the GSM provider and SIM cards with identical tariffs, for example).
   b. Assign the SMS groups created in step 1.a above to the User groups if you use messaging with different routing rules to different company departments, for example.
2. Add users in the **Monitoring–>Users–>Add** menu:
   a. Assign the user to a **User group**.
   b. Set **Access type: Email(SMTP/POP3)**.
   c. Define the **Login name (without domain)** and **Password** to log in to the SMS server via your e-mail client.
3. Set the Domain in the **Messaging–>User Group–>General settings–>Email settings–>Domain** menu.
   a. Set the user domain. The domain is automatically assigned to the user name in the selected User group and represents the user e-mail address. Example: Suppose a user was created in step 2.c with **Login name: USER** and domain **Domain: email.com**, then the e-mail address of the user

with the right to access the SMS server will be **USER@email.com**. The @ character is completed automatically and is not included in the Domain: field.

4. Create the incoming SMS routing rules in the **Messaging–>User Group–>General settings–>Incoming SMS routing–>Add** menu:
   a. Select the group to receive the SMS via the **Message from SMS group**.
   b. Select the **Action type** to define how to process incoming SMS. Example: Set **Action type: Deliver to user** to deliver SMS to selected users.
   c. Set **Deliver to user** to define the SMS receiving user.
5. Create the outgoing SMS routing rules in the **Messaging–>User Group–>General settings–>Outgoing SMS routing–>Add** menu:
   Enable the SMS server in the **Messaging–>Modules grouping–>SMS server enabled** menu.
   a. Set the **SMS group** via which the outgoing SMS will be routed in the **Message to SMS group** menu.
   b. Select the **Action type** to define how to process outgoing SMS. Example: Set **Action type: Submit from user** to send SMS from selected users.
   c. Set **Submit from user** to define the SMS sending user.
6. Should you get in troubles, follow the steps below please:

- Read the User Manual carefully and check all parameters.
- Find answers to the frequently asked questions at http://faq.2n.cz.
- Consult your servicing partner.

You are recommended to attend a 2N certified training to improve your installation chances.

Monitoring

This section helps you configure sending of SNMP traps. Download the list of available traps, i.e. the MIB table, from the gateway web interface. The main purpose of SNMP monitoring is to observe your GSM gateway states continuously and send state information to the defined IP addresses whenever an error occurs.

Refer to the **List of SNMP traps** for the list of all SNMP traps sent by **2N® VoiceBlue NEXT**.

Caution

- This service is subject to licence! Refer to **Management–>Licence key** for the current GSM gateway licence state details.

Common Settings

General

- **Signal level –** define the maximum/minimum level of the signal received by the GSM/UMTS module. The SNMP trap is sent automatically whenever the defined values are exceeded.
- **SMS bufer upper limit** – define the maximum filling capacity of the SMS storage. The SNMP trap is sent automatically whenever the defined value is exceeded.

SNMP

General

- **Enabled** – enable/disable the SNMP service.
- **Community** – SNMP community is a group that owns the device and control stations on which SNMP is running. Default value = public.

Servers

- **First SNMP server** – the addressee's primary IP address and transport port for SNMP traps (default port = 162). Please make sure that the parameters have been completed correctly to avoid SNMP malfunction.
- **Second SNMP server** – the addressee's secondary IP address and transport port for SNMP traps (default port = 162). If these parameters are correctly completed, the SNMP traps will be sent both to the primary and secondary IP addresses.

Download MIB files

This section helps you download the MIB files that are necessary for a proper function of your MIB browser.

Caution
The SNMP service works properly on condition that

- SNMP is licensed in the **Management** section.
- The primary IP address, transport port and community parameters are completed.
- SNMP is enabled in the **Monitoring** section.e

List of SNMP traps

Type	Text
VoIP	VoIP SIP line is not registered
	VoIP SIP line is registered
GSM	First GSM module: ERROR
	First GSM module: OK
	First GSM module has too strong signal level
	First GSM module decreased under level of too strong signal level
	First GSM module has weak signal level
	First GSM module increased over level of too weak signal level
	Second GSM module: ERROR
	Second GSM module: OK
	Second GSM module has too strong signal level
	Second GSM module decreased under level of too strong signal level
	Second GSM module has weak signal level
	Second GSM module increased over level of too weak signal level
	Third GSM module: ERROR
	Third GSM module: OK
	Third GSM module has too strong signal level
	Third GSM module decreased under level of too strong signal level
	Third GSM module has weak signal level
	Third GSM module increased over level of too weak signal level
	Fourth GSM module: ERROR
	Fourth GSM module: OK
	Fourth GSM module has too strong signal level
	Fourth GSM module decreased under level of too strong signal level
	Fourth GSM module has weak signal level
	Fourth GSM module increased over level of too weak signal level
SMS buffer	Number of SMS/SMPP messages in memory exceeds defined max level
	Number of SMS/SMPP messages in memory decreased under defined max level
	Number of SMS/SMPP messages in memory exceeds defined max level
	Number of SMS/SMPP messages in memory decreased under defined max level

Others

SIM Client

This section is displayed for those GSM gateways only that support the **2N® SIM Star** system connection – remote SIM card function.

SMS

This subsection helps you send and receive SMS via the modules.

Tip

- You can also use the SMS user account for SMS sending/receiving. Refer to the Web Configuration subsection for SMS user configuration details.

Utils

This section provides extra functions of the GSM gateway. Currently, they include Ethernet interface tracing (**Network trace**), which helps detect gateway installation problems, such as SIP communication debugging. The downloaded file is in the PCAP format, which is, for example, readable by WireShark (http://www.wireshark.org/).

Management

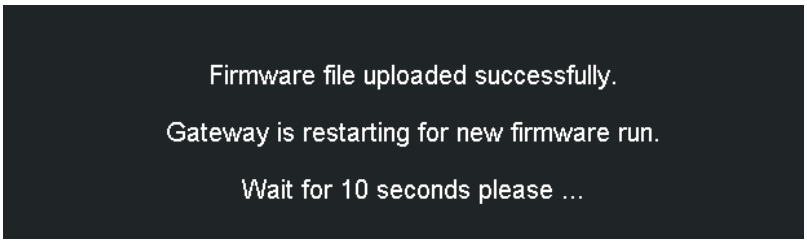
Firmware update

In this menu you can store new firmware into the GSM gateway. Select the **VBN–xx.xx.xx.bin** files only. Refer to the Firmware Upgrade subsection on page Brief Installation Guide for more firmware information.

Warning

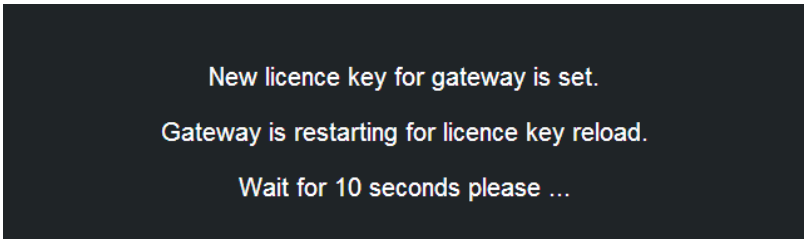
- Use the **firmware certified** for this gateway type only! Any other firmware type may **damage 2N® VoiceBlue Next irreversibly**!

After the new firmware file is stored and sent, the GSM gateway is restarted automatically.



Licence key

This section helps you upload the licence file/code to **2N® VoiceBlue NEXT** and provides information on the current licence file. Refer to Gateway Control for more details.

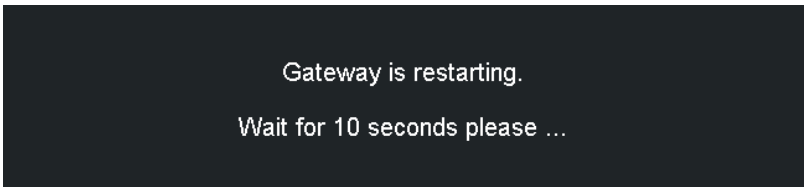


Caution

- By inserting a new licence code you restart the GSM gateway and discontinue all the currently made calls!

Restart

This subsection provides information on a forced restart of **2N® VoiceBlue Next**.



Configuration download

The window helps you download the current **2N® VoiceBlue Next** configuration. The file format is **CFG–M202–gateway_serial_number–rrrrmmdd–hhmmss.tar**.

Configuration upload

The window helps you upload new configuration settings into **2N® VoiceBlue Next** co nfiguration. The Ethernet interface and login data remain the same!

Warning

- Be sure to upload the configuration file intended for the selected GSM gateway only to avoid malfunction of the GSM gateway and the necessity to restart the system.

Configuration Reset

This subsection helps you restore the default configuration (except for Ethernet and login).

- **Including LCR table** – enable/disable keeping of the LCR table records .
- **Including CLIP routing table** – enable/disable keeping of the CLIP routing

table records.

Documentation

Download the User Manual saved in the gateway.

Caution

- The User Manual version will not change during the gateway firmware upgrade. Click on the link to www.2n.cz for the latest User Manual version!

4. Terminal

This section describes communication with **2N® VoiceBlue Next** using a terminal.

Here is what you can find in this section:

- 4.1 Terminal-Based Communication
- 4.2 AT Interface
- 4.3 LOGs
- 4.4 CDR
- 4.5 SDR
- 4.6 Available Status Messages
- 4.7 Statistics
- 4.8 Tracing
- 4.9 External SMS Control

4.1 Terminal-Based Communication

2N® VoiceBlue Next allows for communication via the TCP/IP Telnet protocol. The maximum count of active Telnet sessions is ten. If there are more Telnet connection requests than ten, the GSM gateway rejects the connection.

Terminal

2N® VoiceBlue Next acts as an ANSI terminal with echo. Text commands are entered in one of the following formats: AT[command]<CR>, or AT[command]<CR><LF>. The answer consists of up to several rows, an empty row with <CR><LF> and the reply type: OK, ERROR, or BUSY. Use A/<CR> to repeat the command.

The gateway requests a password by displaying the **Login:** and **Password:** prompts. It uses an extended set of AT commands for configuration. Refer to Subs.4.2 for the list of terminal AT commands.

Example of a login window:

[VoiceBlue Next] V–01.00.01 B–00.91

Date/time: 15.1.2010/20:56:53.98

SNumber: M202–1501270008

Login: Admin

Password: **

Access Data

2N® VoiceBlue Next receives Telnet requests on port 23. Port 23 can not be reconfigured.

The access password and name are identical with those of the web interface. The access data cannot be changed from the gateway Telnet interface (only from the web interface) for security reasons.

Refer to Subs. 2.2 for access data factory settings.

4.2 AT Interface

2N® VoiceBlue Next uses the list of AT commands derived from the standard AT set for configuration and monitoring purposes.

The configuration entering commands start with AT%parameter_id.

The configuration writing out commands start with AT&section.

The following list of AT commands is universal and some commands may not be included in **2N® VoiceBlue Next**.

Basic Commands

System Information	
ATI3	Firmware version and copyright
ATI4	Factory number
Statistics	
AT&T	Statistics of calls via PRI interface
AT&TIN	Statistics of incoming calls via GSM modules
AT&TOUT	Statistics of outgoing calls via GSM modules
AT&T##	Statistics of calls via SIM cards of GSM module ##
AT%TPR=mi,ci,mo,co	Setting of statistics via PRI
AT%TG0=mi,ci,0,0	Setting of statistics of incoming calls via all GSM modules in GSM group 0
AT%TG#=mi,co,ri,ro	Settings of statistics of GSM group #
AT%TGALL=mo.co,ri,ro	Settings of statistics of all GSM groups
AT%T##=#,mi,ci,si,mo,co,so	Settings of statistics of SIM card # of GSM module ##
AT%T##=ALL,mi,ci,si,mo,co,so	Settings of statistics of all SIM cards of GSM module ##
AT%TALL=ALL,mi,ci,si,mo,co,so	Settings of statistics of all SIM cards of all GSM modules
Description of parameters:	Mi/mo > Minutes of incom./outg. callsCi/co > number of connected incom./outg. callsRi/ro > number of re–routed incom/outg. calls in groupSi/so > number of received/sent SMS messages
! When you change these parameters, all the other counters will be reset!!	
Wireless Engines	
AT&Q##	Received GSM signal level (##=00–15)
AT&QALL	Received GSM signal level from all GSM modules
AT&Gxx=atcmd	Send command for AT commands directly to GSM module
	AT&Gxx=at+cnum – SIM card telephone number AT&Gxx=at+cpin="xxxx" – PIN entering AT&Gxx=at+clck="sc",1"xxxx" – PIN request enable AT&Gxx=at+clck="sc",0"xxxx" – PIN request disable

LOG and CDR List	
AT&L	Log file listing (PRIGW awaits pressing of the Space bar after every 20 lines)
AT&C	Call file listing (PRIGW awaits pressing of any key after every 20 lines)
AT&CR	Call file listing and deleting (PRIGW awaits pressing of the # key to delete this line /with call info/ from memory and sends next line)

Configuration commands

Status Information	
AT&V	Overview of all system settings
AT&VI	Overview of ISDN PRI settings
AT&V0	Overview of basic GSM settings
AT&V#	Overview of GSM group # settings (#=1–8)
AT&VALL	Overview of all GSM group settings
AT&N#	Overview of parameters of network list # (#=1–8)
AT&NALL	Overview of parameters of all network lists
AT&R	Overview of LCR lines
Default Reset	
AT&FRES	Reset with gateway factory settings
System Settings	
AT%S70=x.x.x.x	IP address
AT%S71=x.x.x.x	IP address mask
AT%S81=	Initialisation string for setting of mode connected to the gateway COM2
AT%S90=x	Type of assignment of B–channels to GSM modules (0–rotating; 1–locked; 2–according to credit)
AT%S91=x,y	Way of call information recording (0– no record; 1– successfully connected outgoing call; 2– successfully connected incoming calls; 3– successfully connected incoming and outgoing calls; 5– all outgoing calls; 6– all incoming calls; 7–all incoming+outgoing calls) Y = to each call add NET info (1) or no (0)
AT%S93=x	Global selection of SIM card (0= according to GSM group, 1–8= number of SIM holder)
AT%S98=xxxx	PIN code for SIM cards inserted in the gateway
AT%S99= dd.mm.yy.w/hh:mm:ss	Gateway date and time settings
AT%X00=hout,hin,min	Automatic logout of GSM moduleshout ...hour when the modules start to log out randomlyhin ...hour when the modules start to log in randomlymin ...minimum number of modules in a group (2..15) that always remain logged–in
AT%X80=name/pass	Username/password for access via IP port (this parameter is write only– you cannot see its value)

System Control	
AT&BSYS=RESET	Reset of PRIGW
AT&Gxx=RESET	Reset of GSM module xx
AT&Gxx=BLOCK	Block of GSM module xx
AT&Gxx=DOWN	Transfer of GSM module into the sleep mode
AT&Gxx=ON	Transfer of GSM module into the idle mode
Basic GSM Settings	
AT%G00=rr,gggggg,gggggg, gggggg,gggggg,gggggg	g,r=Number of GSM group for each GSM module
AT%G01=mode,atms,atfs	Voice processing settings (for GM22 GSM modules only) mode – algorithm (0–none, 1–semiduplex, 2–fullduplex)atms,atfs – transit, receive gain (3=+5dB, 1=+2.5dB, 0=0dB, 2=–2.5dB, 4=–5dB
AT%G02=mode,atms,atfs	Voice processing settings (for TC35 GSM modules only) mode – algorithm (2–echo canceller)atms,atfs – transit, receive gain (3=+5dBm, 1=+2.5dB, 0=0dB, 2=–2.5dB, 4=–5dB
AT%G06=mmdd,…mmdd…	List of holiday days (mm–month;dd–day), max length of line is 63 chars!
AT%G07=mmdd,…mmdd	Net list of holiday days
AT%G08=delay,min,max,tout	Parameters of dialling from ISDN to GSMdelay–relax delay between callsmin – minimum length of number dialled from ISDNmax – maximum length of number dialled from ISDNtout – delay between dialled digits
GSM Settings (X–GSM Group)	
AT%G#0=sim	Mode of switching SIM card sim=0: 0 – OFF – group is turned-off sim>0: 1-4 – SIM card 1-4; 5 – switching by time; 6 – switching at random time
AT%G#1=netid,clir,min,sms, sec,sec2,pseudo	Basic parameters of SIM card in slot 1 netid–network number clir – 0=default,1=enabled,2=disabledmin – maximum number of minutes called sms – maximum number of SMS sent sec – minimum call length (not for cdr info)sec2 – accuracy of call length measurement (not for cdr info)pseudo – default = 0
AT%G#2=netid,clir,min,sms, sec,sec2,pseudo	Basic parameters of SIM card in slot 2
List of Networks	

AT%N#0=op/np,...op/np	List of prefixes replaced by new prefixes (#–number of netlist; op–old prefix;np–new prefix)
AT%N#1=pr/n,...pr/n	List of prefixes dialled into ISDN (to be used for LCR table) pr=prefix n=length of number (parameter /n is optional)
AT%N#2..7=pr/n,...pr/n	Other 7 lists of prefixes dialled into ISDN
AT%N#9=net,max	Net–network number, max–default length of number dialled from ISDN
LCR Table	
AT&R	Shows LCR table
AT%R##=net,hh:mm/hh:mm/w ,gr*	Settings of line ## (0..63) of LCR tablenet – number of network list (0..8)hh:mm/hh:mm/w* – line using time limitation gr – GSM group numbers
Test Calls	
AT!SGxx=dial	Initiates testing call from AUX interface (xx is the called interface): 00..31 GSM module 32– to GSM via GSM module according to call prefix
AT!WGxx	Connects the following call from interface xx to AUX interface 00..31 from GSM No. 32 first incoming calls from GSM
AT!D	Terminates call on AUX interface
Trace	**Trace can be activated for only one session at a time**
AT!RE	Start tracing of error messages on active interface
AT!Lx	Start tracing of LAN & Telnet on layer xx=2 – Layers IP,TCP,Telnetx=3 – Layers TCP, Telnetx=4 – Layer Telnet
AT!Rx	Start tracing of ISDN PRI & GSM on layer xx=1 – Layers 1,2,3,4x=2 – Layers 2,3,4x=3 – Layers 3,4x=4 – Layer 4
AT!RR	Start tracing (AT!R2) to COM1 interface
AT!RX	Stop tracing (AT!R2) to COM1 interface
SMS Control	**SMS control can be activated for only one session at a time**
AT!G=#	A6 – activate control via used session55 – deactivate control via used session99 – deactivate control on all sessions
AT^MS=ch	Module status ch=0 – 31

***module: ch,sim,sts3,sts2,phs2,type**	Response to AT^MS:ch – number of wireless enginesim – number of currently used SIM: 0–none, 1–8–holder 1–8sts3 – status of layer 3sts2 – status of layer 2phs2 – substatus of layer 2type – type of wireless engine
AT^MI=ch	Module info ch = 0 –31
***minfo: ch,sim,netid,"oper",new,cnt**	Response to AT^MInetid – network ID of active wireless networkoper – name of active wireless networknew – number of received and saved SMS on SIM cardcnt – total number of sent SMS messages according to statistics
***smsinc: ch,ix,sts,mem**	SMS message has been received and savedix – index number of saved SMS (0–250)sts – status of SMS (from +CMGL: ix.sts.len)mem – type of used storage (1,2,3)
***smsrep: ch,ix,sts,mem**	SMS status confirmation has been receivedsts – reserved (0)
***smsout: ch[/id],ref,req**	SMS message was sent and not saved to SIM cardRef – reference number of SMS (0–255) – to be used in status confirmation messagesID – third party identification number of SMS
***smspdu: ch,ix,sts,len,pdu,csum**	Content of SMS messagelen – length of SMS message (number of bytes in PDU)PDU – content of message in PDU formatcsum – checksum of all PDU bytes (2 hexa digits) calculated without carry
***smsdel: ch,ix**	SMS message deletion confirmation
***smserr: ch[/id],ix,req,err,cms**	Fault command response, or SMS not sentreq – required wireless engine or GSM grouperr – error code (6–busy,40–write,41–read,42–delete,43–list)cms – error cause from wireless engine
***smssel: ch,cnt**	SMS storage change confirmation
AT^SX=ch	SMS listing – request for listing of all SMS messages and status confirmations saved on SIM card. Possible answers:: *smserr (busy,list) or *smsinc (ix=1..255) for each saved SMS or status SMS , end of list or empty SIM card – *smsinc (ix=0).
AT^SR=ch.ix	SMS read – request for reading of an SMS message or SMS status saved on SIM card. Possible answers: *smserr (busy,read) or *smspdu
AT^SD=ch,ix	SMS delete – request for deletion of an SMS message (or SMS status message). Possible answers: *smserr (busy,delete) or *smsdel

AT^SM=ch,len,pdu,chsum	SMS to module – request for sending a message via GSM module 0..31 or via any GSM module (ch=32). Possible answers: *smserr (busy,write) or *smsout
AT^SG=grp[/id],len,pdu,csum	SMS to group – request for sending an SMS message via GSM group 1..8. Possible answers: *smserr (busy,write) or *smsout
AT^SS=ch,mem	Command for changing SMS storage. Mem (1;2;3) = type of SMS storageSiemens:1=SM, 2=ME, 3=MEEricsson:1=SM, 2=ME, 3=MEWavecom:1=SM, 2=ME, 3=SRHuawei:1=SM, 2=SM, 3=SMSierraWir.:1=SM, 2=ME, 3=SRMotorola:1=IM, 2=IM, 3=IM

4.3 LOGs

For the current LOG records see the **Gateway / Gateway control – LOG file** section or use the AT&L AT command for Telnet sessions. The following list of LOG statuses is universal and some LOGs may not be included in **2N® VoiceBlue Next**:

Type	Text	Description
POWER	[Power on]	System switched on
	[Power off]	System switched off
	[Warm boot]	Restart of system, unknown cause
	[Watchdog]	Restart of system by watchdog
	[BKPT code]	CPU error: break code detected
	[Stack error]	CPU error: stock integrity failure
	[Divided by zero]	CPU error: dividing by zero
	[RETI code]	CPU error: illegal using of instruction reti
	[NMI intr]	CPU error: wrong interrupt
	[VOID intr]	CPU error: wrong interrupt
	[Upgrade reset]	Start of upgrade firmware procedure
	[Software reset]	Reset by AT commands (at&fres…)
INIT	Eeprom	Initialisation of eeprom (configuration)
	Flash	Initialisation of flash memory (firmware)
HW–ERR		(##…address of chip, RD…read value/WR…expected value)
	Codec ##,RD/WR	Error in initialisation of codec on GSM,AUX board
	COM2 #####,RD/WR	Error in initialisation of COM2 on AUX board
	Duart #####,RD/WR	Error in initialisation of serial controller on GSM board
	Hscx #####,RD/WR	Error in initialisation of HDLC controller on AUX board
	Pri #####,RD/WR	Error in initialisation of PRI controller
SYSERR	User stack error!	SW error: stock integrity failure
BRDIN	#08 TYP STS	Board inserted (number of gsm board, type, status)
BRDOUT	#08 TYP STS	Board disconnected
BRDRES	#08 TYP STS	Reset the board by AT command

	ALL GSM RESET CMD	Reset all GSM boards by AT command
	SYSTEM RESET CMD	Reset the system by at&bsys command
L1–ERR		(reserved)
L2–ERR		Error of ISDN layer 2 by Q.921
	A: unsol.RX(F)	Received packet RR,RNR,REJ with unexpected bit F=1
	B: unsol.DM(F)	Received packet DM with unexpected bit F=1
	C: unsol.UA(F)	Received packet UA with unexpected bit F=1
	D: unsol.UA(–)	Received packet UA with unexpected bit F=0
	E: unsol.DM(–)	Received packet DM with unexpected bit F=0
	F: peer re–est	Received packet SABME in status OKEST
	G: repeated SABME	Repeated unsuccessful send of packet SABME
	H: repeated DM	Repeated unsuccessful send of packet DM
	I: failed (TIMREC)	Unsuccessful breaking–up of status TIMREC
	J: N(R) error	Received wrong value N(R) – numbering of packets
	K: recv.FRMR	Received packet FRMR (information about error)
	L: undef.frame	Received packet of unknown type
	M: (I field)	Received wrong I–packet (numbered packet)
	N: frame size	Received packet with wrong length
	O: N201 error	Value N201 was exceeded (max length of packet)
L3–ERR	tout sts # (p##)	Error of isdn layer 3: timeout in status # on channel p##
G2–ERR	ATD/ERROR init (g##)	Error of isdn layer 2: restart of module g## after rejected command ATD by GSM network
	GSM Cause 150 (g##)	Error of isdn layer 2: restart of module g## after cause 150 was received (call barred by GSM network)
G3–ERR	tout sts # (g##)	Error of isdn layer 3: timeout in status # on module g##
C4–ERR	tout sts # (p##/g##)	Error of connecting layer 4: timeout in status # on call between channel p## and GSM module g##

4.4 CDR

For the current CDRs see the **Gateway–>Gateway control–>CDR file** section or use the AT&C / AT&CR AT commands for Telnet sessions.

****Date Time Type Cause From/to Groups conn. durat. aoc-info gid CDN CGN IMSI/SCID**

- **Column 1**: **
- **Column 2:** call date and time
- **Column 3:** call status
  - I–FD : failed incoming call
  - I–OK : successfully connected incoming call
  - O–FD : failed outgoing call
  - O–OK : successfully connected outgoing call
- **Column 4:** release cause sent to ISDN
- **Column 5:** used channel number / used GSM module
- **Column 6:** used GSM group (C=CallBack)
- **Column 7:** connection time
- **Column 8:** call duration mmm:ss (max. 255:59), or error cause for unconnected call
- **Column 9:** call cost (to be implemented in higher firmware versions) / gateway ID (optional)
- **Column 10:** called party number (CDN)
- **Column 11:** calling party number (CGN)
- **Column 12:** SIM card slot number / SIM card IMSI or SCID

4.5 SDR

Refer to the **Gateway->Gateway control->SDR file** section for the current SDRs.

** **date** **time** **type** **error** **usr/gr/mo** **ssec** **date** **time** **pn/pt** **gid**

- **Column 1**: **
- **Column 2**: date/time – SDR record creation time
- **Column 3**: type of SMS
  - R-OK – delivered OK – SMS received
  - S-FD – sending ERROR – SMS sending error
  - S-OK – send OK – SMS was sent but no SMS delivery receipt was required
  - S-TO – sending timeout – SMS was not sent within the set timeout
  - D-ER – delivery ERROR – SMS delivery error
  - D-OK – delivered OK – SMS delivered
  - D-TO – delivery timeout – SMS was not delivered within the set timeout
- **Column 4**: error type
  - for (D-OK): SMS receipt status (typically E – 000)
  - for (D-ER): SMS receipt error code
  - for (S-FD): module +CMS ERROR (typically E – 028)
  - for (S-FD): internal error (E – 902 = no rule to send)
- **Column 4**: user id/SMS group/GSM module
- **Column 5**: SMS sending time
- **Column 6**: date/time
  - First row – SMS sending time or total sending time for all parts of a long SMS
  - Second row – time of SMS adding to the outgoing/incoming SMS database
- **Column 8**: part number/total
- **Column 9**: gateway id

4.6 Available Status Messages

The following list of statuses is universal and some messages may not be included in **2N® VoiceBlue Next**.

GSM Layer Statuses

GSM Layer 2

Name	Description	What Follows
INIT	GSM module initialisation start	PINREQ
SIM0	Module switch to internal SIM card	INIT
PINREQ	Module PIN request	PINSET
PINSET	PIN value sending to module	INFO
BLOCK	Module blocked temporarily or permanently (see Layer 2 information)	INIT
SET-UP	Module configuration running	IDLE
SLEEP	Module sleep running (transition to BLOCK status)	BLOCK
NWAIT	Awaiting GSM log-in	SET-UP
IDLE	Rest status, can start/receive call, execute AT&G command	CALL/CMD
CALL	Call running (call establishing, connected call, call end)	IDLE
CMD	Communicating with module upon AT&G command	IDLE
INFO	Information loading from module and SIM card	NWAIT

GSM Layer 2 Info for BLOCK

Name	Description
(block)	Blocked by AT&G command, terminated by module, board or system reset
Netw-reg	GSM log-in refused (not activated SIM), next attempt in 5..60 minutes
Clir-err	CLIR activation request refused, next attempt in 2..10 minutes
Mod-err	Defective or unconnected GSM module, next attempt in 2 minutes
Pin-err	Wrong PIN, permanently in BLOCK status
Sim-err	Defective or non- inserted SIM card, next attempt in 2 minutes
Log-out	Temporarily logged-out according to parameter X00
Puk-req	SIM card requires PUK
(undef)	This position of SIM card was not checked by gateway

GSM Layer 3

Name	Description
NULL	Rest status, ready for call
CINIT	Request of call to GSM (from Layer 4)
OPROC	Call to GSM is connected
CDELIV	Call to GSM is ringing
CPRES	Indication of call from GSM (from Layer 2)
CRECV	Call from GSM is ringing on ISDN interface
IPROC	Call from GSM is processed by ISDN interface
ACTIVE	Connected call
DISREQ	Disconnection request (to Layer 2)
DISIND	Indication of disconnection (from Layer 2)
RELREQ	Module release for another call (to Layer 2)
OVRECV	Reading of DTMF dialling from GSM (from Layer 2)
RELIND	Indication of module release (from Layer2)

Link Layer Statuses

Layer 4

Name	Description
NULL	Rest status, ready for call
MORE	Call request received from GSM or ISDN, awaiting further dialling or timeout
SET-UP	Call request sent to opposite interface
PROC	Call request confirmed, awaiting call answer
ACTIVE	Connected call
DISC	Call disconnection in progress

Command Shell Statuses

Command Shell

Name	Description
IDLE	Rest status, awaiting AT commands
REQUEST	Communication with GSM module upon AT&G command is displayed
REPORT	Active tracing
LOGFILE	Content of log file is being written out
CALLFILE	Content of call file is being written out
CALLREAD	Call file records are read
AUTOFILE	Content of incoming call auto routing chart is being written out
QUERY	(not used so far)
SCREEN	(not used so far)
MATRIX	Active matrix screensaver
FUNCFILE	Content of debug file is being written out
LCRFILE	(not used so far)

Telnet Layer

Name	Description
LOGOUT	User logged-out
LOGIN	User is entering name
PASSW	User is entering password
IDLE	User logged-in

Records of Events and Calls

Record Type in Log File

Name	Description
POWER	System power on, power off or reset
INIT	Initialisation of EEPROM (configuration) or Flash (upgrade)
HW–ERR	HW error of plugged-in board
SYSERR	System error (memory error, etc.)
G2–ERR	GSM module layer 2 error (gate2)
G3–ERR	GSM module layer 3 error (gate3)
C4–ERR	Link layer error (call4)

Record Type in Call File

Name	Description
I–FD	Unconnected incoming call attempt
I–OK	Connected incoming call
O–FD	Unconnected outgoing call attempt
O–OK	Connected outgoing call

4.7 Statistics

The GSM gateway automatically generates statistic data on all outgoing and incoming calls. These data can be deleted on the user level or though configuration (automatic deletion on a selected day).

Explanation of columns

Pri/grp – type of calls

Reset – date of last reset the statistics

Minutes – number of minutes Hhhh:mm:ss : same number converted to time

Calls – number of calls

SMS – number of sent SMS messages

Reject – number of unconnected calls (no available free GSM module–call rejected with cause 41(42))

Failed – number of unconnected calls (rejected by GSM network)

C.offs – number of unconnected calls (terminated by calling party)

Errors – number of unconnected calls (wrong requests – disallowed prefix, etc.)

Red.in – number of connected calls (rerouted to this GSM group)

Redout – number of connected calls (rerouted to another GSM group)

Statistics of outgoing and incoming groups

[Statistics of calls in groups]

group (reset) minutes hhhh:mm:ss calls reject failed c.offs errors

---

#i1 inc (2.01) 0 0:00:00 0 0 0 0 0

#i2 inc (2.01) 0 0:00:00 0 0 0 0 0

group (reset) minutes hhhh:mm:ss calls reject failed red.in redout

---

#g1 out (2.01) 0 0:00:00 0 0 0 0 0

#g2 out (2.01) 0 0:00:00 0 0 0 0 0

Statistics of incoming calls on all GSM modules

[Statistics of incoming calls on all modules]

network (reset) minutes hhhh:mm:ss calls reject failed c.offs errors

---

#vi inc (0.00) 0 0:00:00 0 0 0 0 0

#grpinc (0.00) 0 0:00:00 0 0 0 0 0

```
group   (reset)    minutes  hhhh:mm:ss   calls  reject  failed  c.offs  errors
------------------------------------------------------------------------
---------
#i1 inc ( 2.01)        0    0:00:00      0      0       0       0       0
#i2 inc ( 2.01)        0    0:00:00      0      0       0       0       0

module  (reset)    minutes  hhhh:mm:ss   calls  reject  failed  c.offs   smses
------------------------------------------------------------------------
---------
#m0 inc ( 2.01)        0    0:00:00      0      0       0       0       0
#m1 inc ( 2.01)        0    0:00:00      0      0       0       0       0
```

Statistics of outgoing calls on all modules

```
[Statistics of outgoing calls on all modules]
network (reset)    minutes  hhhh:mm:ss   calls  reject  failed  c.offs  errors
------------------------------------------------------------------------
---------
#vi out ( 0.00)        0    0:00:00      0      0       0       0       0
#grpout ( 0.00)        0    0:00:00      0      0       0       0       0

group   (reset)    minutes  hhhh:mm:ss   calls  reject  failed  red.in  redout
------------------------------------------------------------------------
---------
#g1 out ( 2.01)        0    0:00:00      0      0       0       0       0
#g2 out ( 2.01)        0    0:00:00      0      0       0       0       0

module  (reset)    minutes  hhhh:mm:ss   calls  reject  failed  c.offs   smses
------------------------------------------------------------------------
---------
#m0 out ( 2.01)        0    0:00:00      0      0       0       0       0
#m1 out ( 2.01)        0    0:00:00      0      0       0       0       0
```

Statistics of calls on specific GSM modules

```
[Statistics of calls on module #0]
sim/dir net/grp    minutes  hhhh:mm:ss   calls  reject  failed  c.offs   smses
------------------------------------------------------------------------
---------
#1  inc     /1         0    0:00:00      0      0       0       0       0
------------------------------------------------------------------------
---------
#1  out     /1         0    0:00:00      0      0       0       0       0
```

```
[Statistics of calls on module #1]
sim/dir net/grp    minutes  hhhh:mm:ss   calls  reject  failed  c.offs   smses
-------------------------------------------------------------------------
---------
#1  inc     /1         0     0:00:00      0      0       0       0       0
-------------------------------------------------------------------------
---------
#1  out     /1         0     0:00:00      0      0       0       0       0
```

4.8 Tracing

2N® VoiceBlue Next allows for operation tracing (VoIP, SIP, GSM). Tracing is accessible in one session only at a time, the other tracing requests are rejected with BUSY. With regard to a rather extensive volume of data, we recommend that the **Write communication into file** function should be activated before tracing. Any Telnet–supporting terminal (HyperTerminal, Putty, etc.) can be used.

Tip

- You are advised to synchronise your GSM gateway with the SIP Proxy, or a PC before tracing.

AT Command	Description
AT%S92=x	Set the tracing details. Refer to the table below for value X.
AT!R2	Start tracing via the active session.
Any character	Terminate tracing.

Layer 3 and 4 Status Listing	Timestamp Write–Out	SMP Packet Write–Out	Layer 2 Status Listing	LCR Listing
1	**2**	**4**	**8**	**16**

The resultant value X is a sum of the required functions (X= timestamps+LCR = 16+2 = 18, e.g.).

4.9 External SMS Control

If this function is enabled for the active Telnet session, the GSM gateway automatically sends information on an incoming SMS message. An SMS message can also be sent using special commands.

Command	Description
AT	'basic keep alive' command (answer 'OK')
AT!G=A6	Activation of SMS control(can be activated only in one session at same time)
AT!G=A55	Deactivation of SMS control
AT^MS=ch	GSM module [ch] status
AT^MI=ch	GSM module [ch] info
AT^SX=ch	Request to list all SMS messages and status confirmations saved in the SIM card / GSM module. Possible answers: *smserr(busy,list) or *smsinc (ix=1–255) for each saved SMS or status SMS. End of list or empty storage – *smsinc(ix=0), or *smserr 255
ST^SR=ch,ix	Request to read an SMS message or SMS status saved on the SIM card. Possible answers: *smserr (busy,read) or *smspdu
AT^SD=ch,ix	Request to delete an SMS message (or SMS status message). Possible answers: *smserr (busy,delete) or *smsdel
AT^SM=ch,len,pdu,csum	Request to send a message via GSM module 0..31 or via any GSM module (ch=32). Possible answers: *smserr (busy,write) or *smsout
AT^SG=grp[/id],len,pdu,csum	Request to send an SMS message via GSM outgoing group 1..8. Possible answers: *smserr (busy,write) or *smsout
AT^SS=ch,mem	Command for an SMS storage change. Ch (0–31) = module number. Mem (1;2;3) = SMS storage type

SMS Storages	
Siemens / Cinterion:	1=SM, 2=ME, 3=ME
Ericsson:	1=SM, 2=ME, 3=ME
Wavecom:	1=SM, 2=ME, 3=SR
Huawei:	1=SM, 2=SM, 3=SM
SierraWir.:	1=SM, 2=ME, 3=SR
Motorola:	1=IM, 2=IM, 3=IM

(SM = SIM card, ME = module, SR = status report memory, IM = inbox memory)

Information Messages

Message	Description
*smsinc: ch,ix,sts,mem	An SMS message was received and saved into the SMS storage.
*smsrep: ch,ix,sts,mem	An SMS status confirmation was received.
*smsout: ch[/id],ref,req	An SMS message was sent and not saved into the SMS storage.
*smspdu: ch,ix,sts,len,pdu,csum	Content of SMS message or status confirmation.
*smsdel: ch,ix	An SMS message or status confirmation was deleted from position ix.
*smserr: ch[/id],ix,req,err,cms	Response to an error command.
*smssel: ch,cnt	Confirmation of an SMS storage change.

5. Technical Parameters

Two-Channel Version

Dimensions	
Dimensions (W x H x D)	170x130x45 mm
Power Supply	
Type	External power adapter (SELF type) 90–230V,50–60Hz to 12V DC 2A Power over Ethernet (PoE 802.3af)
Connector	DC Jack 2.1mm
Power input	Up to 20W
Interface	
Protocols	TCP, UDP, IP, RTP, TELNET, http, DHCP
Connector	1x 10/100BaseT Ethernet
Administration	AT commands, web interface
VoIP	
Signalling	SIP
DTMF	RFC2833
Count of voice channels	2
Voice codecs	G.711 PCM at 64 kbps G.729 Annex AB
PCM companding	A–law/u–Law (optional)

Wireless Interface	
Count of channels	2
Bandwidth	GSM 850/900/1800/1900 MHz UMTS 800/850/900MHz/2100 MHz
GSM	Cinterion MC55i, or Wavecom Q55, or Telit* UMTS
Audio	HR/FR/EFR/AMR
USIM/SIM card	Small plug-in 3V
Antenna	One external antenna (integrated antenna splitter) SMA FEMALE connector
Others	
Working temperature	5°C – 45°C
Air humidity	Max 95% at 40°C

* according to the GSM module type used

Four-Channel Version

Dimensions	
Dimensions (W x H x D)	220x45x140 mm
Power Supply	
Type	External power adapter (SELF type) 90–230V,50–60Hz to 12V DC 2A Power over Ethernet (PoE 802.3af)
Connector	DC Jack 2.1mm
Power input	Up to 20W
Interface	
Protocols	TCP,UDP,IP,RTP,TELNET,http,DHCP
Connector	1x 10/100BaseT Ethernet
Administration	AT commands, web interface
VoIP	
Signalling	SIP
DTMF	RFC2833
Count of voice channels	4
Voice codecs	G.711 PCM at 64 kbpsG.729 Annex AB
PCM companding	A–law/u–Law (optional)

Wireless Interface	
Count of channels	4
Bandwidth	GSM 850/900/1800/1900 MHz UMTS 800/850/900MHz/2100 MHz
GSM	Cinterion MC55i, or Wavecom Q55, or Telit* UMTS
Audio	HR/FR/EFR/AMR
USIM/SIM card	Small plug-in 3V
Antenna	One external antenna (integrated antenna splitter) SMA FEMALE connector
Others	
Working temperature	5°C – 45°C
Air humidity	Max 95% at 40°C

* according to the GSM module type used

6. Supplementary Information

This section provides supplementary information on **2N® VoiceBlue Next**.

Here is what you can find in this section:

- 6.1 Troubleshooting
- 6.2 List of Abbreviations
- 6.3 Regulations and Directives
- 6.4 General Instructions and Cautions

6.1 Troubleshooting

For the most frequently asked questions refer to faq.2n.cz.

- No LED is shining on **2N® VoiceBlue Next**.
  - **2N® VoiceBlue Next** is disconnected from the power supply.

  2N® VoiceBlue Next fails to log in to the GSM network.
  - Check the SIM card.
  - Check the PIN.
  - Check the antenna connection.
  - Select a place with a good GSM signal.

  2N® VoiceBlue Next is not communicating with the PC.
  - Check the Ethernet cable connection and the Ethernet interface settings in your PC.

6.2 List of Abbreviations

Following abbreviations are used within manual:

- **API** (Application Programming Interface)
- **ASR** (Answer Seizure Ratio)
- **BIOS** (Basic Input–Output System)
- **CD** (Compact Disc)
- **CDR** (Call Data Record)
- **CLIR** (Calling Line Identification Restriction)
- **COM** (PC serial interface)
- **CPU** (Central Processor Unit)
- **DHCP** (Dynamic Host Configuration Protocol)
- **DNS** (Domain Name Server)
- **DTMF (**Dual Tone Multifrequency)
- **eCPU** (enhanced CPU)
- **FW (**Firmware)
- **GMT** (Greenwich Mean Time)
- **GSM** (Group Switched Mobile system)
- **GPRS** (General Packet Radio Service)
- **GW** (Gateway)
- **HW** (Hardware)
- **IMEI** (International Mobile Equipment Identity)
- **IMSI** (International Mobile Subscriber Identity)
- **IP** (Internet Protocol)
- **IVR** (Interactive Voice Response)
- **LAN** (Local Area Network)
- **LED** (Light Emitting Diode)
- **NAT** (Network Address Translation)
- **PC** (Personal Computer)
- **PCB** (Printed Circuit Board)
- **PCM** (Pulse–code modulation)
- **PIN** (Personal Identification Number)
- **PRI** (Primary Rate Interface)
- **PUK** (Personal Unblocking Key)
- **SB** (SIM Board)
- **SC** (SIM Client)
- **SIM** (Subscriber Identity Module)
- **SIP** (Session Initiation Protocol)
- **SMS (**Short Message Service)
- **SSH** (Secure Shell)
- **SW** (Software)
- **TCP** (Transmission Control Protocol)
- **UCMD** (UDP commands)
- **UDP** (User Datagram Protocol)
- **UMTS** (Universal Mobile Telecommunication System)
- **UPS** (Uninterruptible Power Supply)
- **USSD** (Unstructured Supplementary Service Data)
- **VoIP** (Voice over Internet Protocol)
- **WAN** (Wide Area Network)

6.3 Regulations and Directives

Europe

2N® VoiceBlue Next conforms to the following directives and regulations:

Directive 1999/5/EC of the European Parliament and of the Council, of 9 March 1999 – on radio equipment and telecommunications terminal equipment and the mutual recognition of their conformity

Directive 2006/95/EC of the European Parliament and of the Council of 12 December 2006 on the harmonisation of the laws of Member States relating to electrical equipment designed for use within certain voltage limits

Directive 2004/108/EC of the Council of 15 December 2004 on the harmonisation of the laws of Member States relating to electromagnetic compatibility

Commission Regulation (EC) No. 1275/2008, of 17 December 2008, implementing Directive 2005/32/EC of the European Parliament and of the Council with regard to ecodesign requirements for standby and off mode electric power consumption of electrical and electronic household and office equipment

Directive 2011/65/EU of the European Parliament and of the Council of 8 June 2011 on the restriction of the use of certain hazardous substances in electrical and electronic equipment

Regulation (EC) No. 1907/2006 of the European Parliament and of the Council of 18 December 2006 concerning the Registration, Evaluation, Authorisation and Restriction of Chemicals (REACH), establishing a European Chemicals Agency, amending Directive 1999/45/EC and repealing Council Regulation (EEC) No. 793/93 and Commission Regulation (EC) No. 1488/94 as well as Council Directive 76/769/EEC and Commission Directives 91/155/EEC, 93/67/EEC, 93/105/EC and 2000/21/EC

Directive 2002/96/EC of the European Parliament and of the Council of 27 January 2003 on waste electrical and electronic equipment.

Industry Canada

This Class A digital apparatus complies with Canadian ICES-003. / Cet appareil numérique de la classe A est conforme a la norme NMB-003 du Canada.

FCC

NOTE: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at his own expense.

Changes or modifications not approved by the responsible party could void the user's authority to operate the equipment.

6.4 General Instructions and Cautions

General Instructions and Cautions

- Please read this User Manual carefully before using the product. Follow all instructions and recommendations included herein.
- Any use of the product that is in contradiction with the instructions provided herein may result in malfunction, damage or destruction of the product.
- The manufacturer shall not be liable and responsible for any damage incurred as a result of a use of the product other than that included herein, namely undue application and disobedience of the recommendations and warnings in contradiction herewith.
- Any use or connection of the product other than those included herein shall be considered undue and the manufacturer shall not be liable for any consequences arisen as a result of such misconduct.
- Moreover, the manufacturer shall not be liable for any damage or destruction of the product incurred as a result of misplacement, incompetent installation and/or undue operation and use of the product in contradiction herewith.
- The manufacturer assumes no responsibility for any malfunction, damage or destruction of the product caused by incompetent replacement of parts or due to the use of reproduction parts or components.
- The manufacturer shall not be liable and responsible for any loss or damage incurred as a result of a natural disaster or any other unfavourable natural condition.
- The manufacturer shall not be held liable for any damage of the product arising during the shipping thereof.
- The manufacturer shall not make any warrant with regard to data loss or damage.
- The manufacturer shall not be liable and responsible for any direct or indirect damage incurred as a result of a use of the product in contradiction herewith or a failure of the product due to a use in contradiction herewith.
- All applicable legal regulations concerning the product's installation and use as well as provisions of technical standards on electric installations have to be obeyed. The manufacturer shall not be liable and responsible for damage or destruction of the product or damage incurred by the consumer in case the product is used and handled contrary to the said regulations and provisions.
- The consumer shall, at its own expense, obtain software protection of the product. The manufacturer shall not be held liable and responsible for any damage incurred as a result of the use of deficient or substandard security software.
- The consumer shall, without delay, change the access password for the product after installation. The manufacturer shall not be held liable or responsible for any damage incurred by the consumer in connection with the use of the original password.
- The manufacturer also assumes no responsibility for additional costs incurred by the consumer as a result of making calls using a line with an increased tariff.

Electric Waste and Used Battery Pack Handling

Do not place used electric devices and battery packs into municipal waste containers. An undue disposal thereof might impair the environment!

Deliver your expired electric appliances and battery packs removed from them to dedicated dumpsites or containers or give them back to the dealer or manufacturer for environmental–friendly disposal. The dealer or manufacturer shall take the product back free of charge and without requiring another purchase. Make sure that the devices to be disposed of are complete.

Do not throw battery packs into fire. Battery packs may not be taken into parts or shirt–circuited either.

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